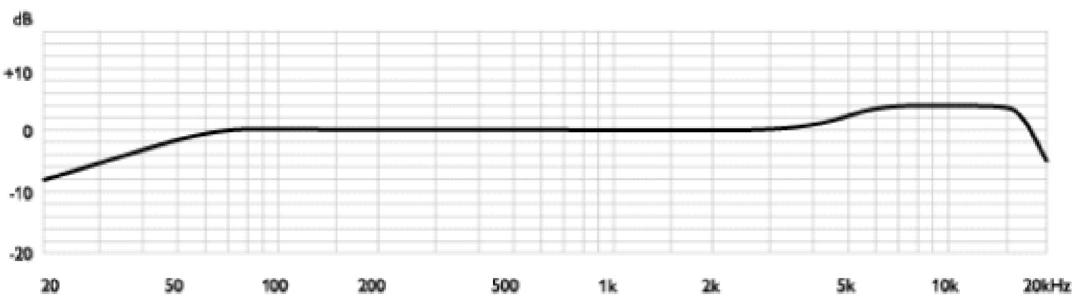
PART I
(Microphone Theory)

Harvey Gerst:

Ok, you're on. At 64, maybe the best thing I can do with my life is to pass on what I've learned from great people that taught me when I was starting out. I think that's why Al Schmitt, George Massenburg, Ed Cherney, and some of the other really big guns spend so much time on the net. We all owe the guys that came before us a lot, and this is our way of paying them back. And that's the only thing I have in common with all those guys I just mentioned - we all kinda drank from the same well back in the 50s and 60s.

I'll try to cover as much ground as I can, to give everybody a good basic understanding of the different mic designs, advantages and disadvantages of each design, how mic polar patterns are created, advantages and disadvantages of each polar pattern, and finally where each type might be used, along with advantages and disadvantages of each usage. How's that for a course outline?



This is the frequency response curve of a Neumann TLM-103. Not very flat, is it? Does that mean it's a bad mic? Before we can answer that we hafta know how to read one of these curves and how to interprete it.

Okay, let's start this with some interesting history as a prelude to the whole mic discussion. "Why" will become pretty clear by the third or fourth paragraph:

In a way, the history of microphones and sound all started with Alexander Graham Bell, and Western Union. After Bell won the lawsuit with Western Union over the invention of the telephone, his fledgling AT&T company needed somebody to manufacture phones for them. Western Union had created a manufacturing division (Western Electric) to make telegraph keys and telegraph equipment. Bell bought the Western Electric division and they had the exclusive right to manufacture phones for Bell.

By 1910, Western Electric had the ambitious task of creating a coast to coast telephone hookup to tie in with the opening of the Panama Canal, but the problem of amplifying a signal over long distances was still unsolved. In 1913, Dr. Harold Arnold (of Western Electric's research group) saw that Dr. Lee DeForest's "Audion vacuum tube" was the possible solution, and they bought the rights to it and began work on a "high vacuum" tube.

This indeed solved their long distance problem, and led to another discovery - a "loud-speaking telephone". In 1916, they received a patent for what we now call a "loudspeaker". With the addition of the "high vacuum" amplifying tube, and another little patent for a device called a "condenser mic", they were suddenly in the P.A. business as well.

These inventions opened the door for radio, talking movies, and sound systems in general, and with their other patent for a high quality "amplifier" in 1916, they pretty much defined the science of sound. (It would be another 12 years (1928) untill a young Georg Neumann would start his own mic company in

Germany. That same year, Western Electric received a patent for a "dynamic mic" design.

The designs Western Electric developed for movie speakers would eventually start companies like Altec and JBL making horns and loudspeakers for Western Electric, and eventually those Western Electric designs became the foundation for their own speaker lines. Western Electric created their own Research and Development arm called "Bell Laboratories", which went on to create the transistor and a host of audio related products. It was Western Electric and Bell Laboratories who we must thank for the development and research into microphone design that we enjoy today.

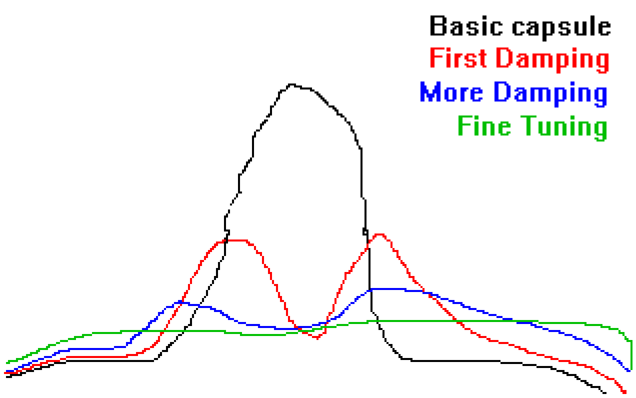
Next, we'll look at some of the different types of microphone designs in terms of advantages and disadvantages. How a "dynamic" mic really works will definitely surprise you (hint: it's NOT just a small speaker in reverse).

Dynamic Mics

By far, the most popular mic on the market today is the dynamic cardioid mic, so that's as good a place as any to start. "How does it work, what exactly is a cardioid, and how and where would you use it" will be our focus today. Let's look inside one and see what we find:

Well, it has a cone (like a small speaker), a voice coil (like a small speaker), and it sits in a magnetic gap (like a small speaker), so isn't it just a small speaker in reverse? Yes, and no. The operating principle is the same, but the execution is very different. When's the last time you saw a 3/4" speaker that went down to 30 or 40 Hz? Here's how it's done:

The system resonance is chosen for a mid band frequency. By itself, the capsule's response looks something like this: (the black line)



just one big resonant peak, with the response falling off rapidly on each side of the peak. Now you can tame that peak by putting in a resonant chamber that's tuned to that peak, which will give you two smaller peaks on either side, like red line:

And if you add two more resonant chambers, tuned for each or those peaks, you wind up looking more like the blue line and finaly the green line:

but remember, it's still a lot like a bunch of tuned coca cola bottles inside there. Now ya gotta do all of this stuff JUST to get the response usable - never mind about the mic pattern yet!

A lot more to come!! Everybody still with me at this point? Any questions?

Chris F :

1. So, if I understand your first post, you explain that the frequency response of your basic garden variety dynamic mic is not really a curve, but rather a series of mechanically engineered peaks, right? But we don't necessarily hear it that way, because our ears/brains fill in the sonic spaces the same way our eyes/brains do when we look at a newspaper photo that really consists of a bunch of dots rather than an actual picture. Is that pretty close?
2. Is the reason for that the size of the diaphagm? It would make sense that, in order to truly reproduce a sound in the extreme low register, the diaphragm would need to be as large as the soundwave corresponding to the lowest note on the recording, which would be both incredibly impractical and terribly funny....can you see Roger Daltrey swinging one of those suckers around? So instead of that, the initial response peak is spread out so that it covers more range more evenly.

   Am I close, or did I get off track by assuming too much?

Harvey Gerst :

1. Yes and no. The broad band resonators and filters actually do smooth out those peaks pretty well, but you hafta remember it's all done with mechanical tricks and if you hit it with enough energy in a susceptable frequency range, it will resonate.

2. No, that's also a function of excursion and mic design. Small omnis for example, can get down to 1 Hz fairly flat.

A little too much assumption which I'll try to explain in the next installment.

Blinddogblues:

Harvey, what I would really like to see is some help on mic selection and placement for various instruments. Especially what I consider hard to mic instruments like piano, acoustic guitar, etc...

Harvey Gerst:

That's coming, but it's important to learn HOW a mic does what it does, so you can understand it's limitations and best uses. Once that's understood, the rest is pretty much, "Well duh, of course!" And you'll be able to say with authority, "No, a large diaphragm mic does not go lower in frequency than a small diaphragm mic", and you'll understand why.

And that will lead you to "when should I use this mic or that mic?", except by that time, you'll know the answer yourself, and beautiful women will flock to you at parties, men will whisper in awe as you walk by, and young girls will seek your autograph, all because YOU understand how mics work and how to use them.

Seriousturtle:

Harvey, i looked at your website today. is it normal to have a zillion microphones? just wondering. you know, if you're throwing any out in the trash, i could take them off your hands....

Harvey Gerst:

Actually, no, it's not normal, but since I'm a bottom-feeder, it's cheaper for me to have a lot of low cost mics around for different colors than it is to have a lot of expensive mic pres - for different colors. Most of the mics I bought very cheap, at pawn shops, garage sales, ebay, newspaper ads, etc. I'm still paying on a few of the more expensive ones. At retail, it looks impressive, but I've never paid more than $50 for a Shure SM-57, way under $100 each for the Sennheiser 421s, etc.

My only "bought brand new" mics are my Coles, the Neumann TLM-103, the Oktavas, and the Marshalls - everything else was purchased used, for very cheap. And I actually traded some old stuff I had for the Neumann.

Seriousturtle :

Do you like the sound of the TLM-103 for vocals, which is what i assume you use it for? how much different does it sound than the Ui 87? keep preaching about the response of the mic, cuz i'm interested in how an unflat response is considered good. since i hear crap about how you should always go for the flat response.

Harvey Gerst :

Wow, difficult to answer, but I'll try.

I've used the 103 for vocals, acoustic guitar, mandolin, fiddle, and a host of other instruments. It's a very clear sounding mic with a nice mid range warmth, and no shrillness. I'm not a big fan of the U87 but a lot of people love them.

Finally, "flat" is hard to define. The TLM103 is flatter than a U87, but that's not the point. You want to go for the most flattering mic for a particular instrument, but it must be a mic that doesn't add unpleasant coloration.

The flattest mics are/were made by B&K for test measurements and they're ruler flat (literally) usually from about 10 Hz to around 30 or 40 kHz. They are also pretty boring as mics for recording most music.

Harvey Gerst :

Part 3 (or 4 or whatever)

Well, I haven't heard from David Satz as far as permission to reprint his post here, but I don't think he'll mind (since he's a nice guy), so here it is:

RockyRoad wrote:

Could some kind person explain to me how the physics of these things work, and how sound from behind an omni mic such as the KM183 can get around the metal side casing and into the mic.

Sorry for the dumb questions, but I'd like to know why things arent as they seem on the surface.

"Why are things not the way they seem?" is a question that I _so_ wish people would ask more often than they do. Most folks seem to stop noticing that things aren't the way they seem, and start behaving as if that appearances are all that matter. To me that's the essence of that form of spiritual death which we in this society call "adulthood." It's why I believe that only children should be allowed to vote or own property--but failing that, there should be a law (or better yet, a general agreement) that grown-ups ought to answer all honest questions honestly. Then maybe we would not be such a culture of deception and self-deception, and people would retain their ability to notice things that don't make sense.

The replies from Sean and Scott are spot on, but I'd like to try to help you visualize what these two types of microphone are doing. Again, the relevant categories are "pressure transducer" (basically omnidirectional) and "pressure gradient transducer" (basically figure-8, but by using dual diaphragms and other tricks, any other first-order directional pattern can be synthesized including cardioid and super- or hypercardioid).

The model of a pressure transducer is a barometer. It measures air pressure in the space around it. The simplest, grade-school science barometer is a sealed tin can with air in it. The lid of the can will flex in proportion to air pressure changes in the room around it; you can attach a stick to the lid, and calibrate the stick's motions in terms of whatever units of air pressure you want to use (inches of mercury or the standard metric unit, which is "bars").

The thing is, the can will get squeezed by increasing air pressure or it will expand in times of low air pressure, regardless of which way you "aim" it. In fact the concept of "aiming" a barometer doesn't really exist because it's integrating and responding to a phenomenon that is all around it. You just set it up in whatever physical orientation is convenient for you, and it works.

You could think of the barometric pressure in a daily weather report as being the response of the barometer at 0.000011574 Hz if you want (one cycle per day). Essentially a barometer is a microphone with response down to DC. And that is a real-world characteristic of pressure transducers: their low-frequency response can be extended as far down as you like. Most pressure microphones have some small vent built in to prevent them from bursting when transported by air, but they can very well be dead flat to below 1 Hz or 5 Hz, certainly to any audible frequency.

OK. So the pressure transducer works precisely _because_ only one side of the diaphragm (the lid of the can) is exposed to the air pressure that is to be recorded; the air on the other side of the diaphragm is a constant mass, and the diaphragm flexes in order to equalize the pressure on both its sides.

The other major category of transducer is pressure-gradient, which is a fancy way of saying that its diaphragm is exposed to the sound field both on the front and the back, so it responds to the difference between the pressure that exists on the front and the pressure on the back. If the pressure presented on both sides at a given moment is identical, there is no net motion and no output. If the pressure on the front is greater than the pressure on the back, the diaphragm will move toward its backplate (assuming a condenser microphone). If the opposite is true, the diaphragm will move outwards, away from the backplate.

The thing is, if you just hang a microphone diaphragm out in space, it will be pushed around by wind or by air currents of any kind (including if you just blow on it) but it won't pick up much in the audio frequency band because it's a thin element and the pressure from sound waves will tend to be identical on both sides of the diaphragm, at least until you get up to the high frequencies (which we'll talk about some other day), and when the pressure is the same on both sides of the membrane there is no net movement and no output. But before I explain why this type of arrangement picks up sound at all, let's observe that we've actually encountered something that is true of pressure gradient microphones generally, which is that they are much more sensitive to wind, breath noise and "popping" of consonants in vocal pickup than their omnidirectional counterparts are (when the omnis are pressure transducers).

The trick which makes a pressure-gradient arrangement work for recording sound is that the sound reaching the back of the membrane is delayed momentarily, by setting up a delay chamber in between the back vents of the microphone and the back of the diaphragm. If you can make the pathway for sound even just a tiny fraction of an inch longer before the sound reaches the rear of the diaphragm, then you will cause a phase shift between the sound reaching the front and the sound reaching the back. That phase shift will be different at different frequencies, of course, so there will really be only one frequency (plus its exact integer multiples) at which a maximum of difference in pressure will result between the front and back of the diaphragm. At that frequency the resulting microphone will have its highest sensitivity to sound. But if you arrange things so that this frequency occurs somewhere other than at the very top or the very bottom of the audio range, you can do other tricks with damping and filtering so as to flatten the overall response.

The thing is, this more complicated type of microphone is also sensitive to the direction from which sound is arriving, because if sound is arriving from in front, it will strike the front of the diaphragm immediately, then when it reaches the rear input ports it will pass through the acoustic delay chamber and eventually reach the back of the diaphragm--so there will be a continually varying difference in the air pressure on the two sides of the diaphragm, and that's what moves it and produces a signal. But if the sound is coming from behind the microphone, it will reach the back inlets first, and pass through the delay chamber at the same rate of speed as the original wave is traveling outside the microphone; by the time both waves reach the two sides of the diaphragm, they will be in phase with one another and the result is no net motion of the diaphragm. (That's if the microphone is a single-diaphragm cardioid.)

That should be enough to establish a basic viewpoint, I hope. (End of David Satz' post) And we'll now head into "when and why to use what, and how" when we do the next posting on this.

Chris:

Whew! That's some pretty heavy stuff....it'll take a while to digest. This was my second pass through, and I have a question:

1. if cardiod condensers only receive the sound wave from the front of the diaphragm, why are they all have "vents" (don't know if that's the correct term or not) **behind** the diaphragm? Wouldn't that be superfluous? Maybe I missed something. Or, maybe it's too damn late for me to be trying to comprehend this stuff...I'll try again in the morning.

Great post though. I hope I'm not too dense to get it.

Harvey Gerst :

Read the very last paragraph again. Cardioids work **because** they have vents that let sound into the back side of the diaphragm.

Chris F :

That's what I get for reading something complicated at 2:14 A.M. Okay, in a minuteI'm gonna start to be afraid to ask any more questions since I'm 0 for 2 so far, but until then I'll assume that we're working under the "there are no dumb questions" mindset and plunge ahead. Let me try again, and please enlighten me (again) if I'm wrong. So far, I understand that:

Pressure **transducer**=omnidirectional
Pressure **gradient** = condenser patterns such as figure 8, cardioid, hypercardioid etc.....

If I'm understanding this correctly, condenser mics need to be powered with "phantom power" because the pressure that they are receiving on the front of the diaphragm is balanced by pressure on the back of the diaphragm. The difference in pressure - caused by a "phase delay" that can be measured in microseconds - between the front and the back of the diaphragm is very small, and for this reason, the signal needs to be "amplified" by an electric charge.

Further, the directional "pickup pattern" is determined by the design of the diaphragm capsule - more to the point, the pattern is determined by the amount of "delay" engineered in to the **back** of the capsule. Because if an equal amount of pressure reaches both the front and the back of the diaphragm at the same time, it won't move at all and there will be no sound picked up from the direction that caused this to happen. But sounds coming from any direction that causes the diaphragm to have more pressure on one side or the other **will** be picked up because they're slightly "out of phase".

Am I getting any closer? I have some other questions but I'll hold them until I feel like I undstand this issue better

Sorry if I seem dense, I'm just trying to fully understand some of the basic concepts before you move on to the REALLY DEEP stuff. Could you give a couple of examples of some common industry standard mics which relate to the pressure transducer vs. gradient schism?

George :

A pressure gradient mic depends on delays getting to the back of the diaphragm, wheter it's a ribbon mic, a dynamic moving coil mic, or a condensor mic.

Two things:

a) I don't think I understand the difference between "Dynamic" and "Condensor" mic designs other than the fact that Condensor mics are powered and Dynamics are not. I have noticed that dynamics are often used for applications where the mic is going to be exposed to really high SPL's - like on snare drums played by musclebound rockers using tree trunks for sticks (ex - SM57), kick drums played by same (AT?

25 - the little stubby one), and diva singers (that was meant as a non gender-specific term BTW) belting out their dramatic sh*t with the mic halfway down their throat during live shows (SM58). I think I must have been mistakenly confusing "Dynamic" with what you describe as "Pressure Transducer". Until now, I always thought of condensor mics as being much more of an "ambient" sounding mic.

b) I just figured out during your last post that the word "Condensor" is spelled with an "or" at the end instead of an "er". DOH! Better late than never.

A condensor mic works by the difference in voltage between the back plate and the diaphragm. The voltage can be either a permanent pre-charged voltage (an electret), or a capsule that has 48 volts across one side or more (some B&K mics use over 100 volts to charge the capsule). The phantom power is only one way to get a condenson mic to work - it has nothing to do with the patterns.

Gotcha.

Other than that, you've got it!!! The delay from sound hitting the back side of the diaphragm of any mic results in the different patterns. If the back side of the diaphragm is sealed, it's strictly a pressure mic, and it's omni - always!

Harvey Gerst :

By George, you've **almost** got it!! Forget the condensor/phantom power part and you've got it. A pressure gradient mic depends on delays getting to the back of the diaphragm, whether it's a ribbon mic, a dynamic moving coil mic, or a condensor mic.

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Very good!!! You also asked for some examples of different patterns:

Pressure (omni) - all calibration mics, Earthworks.

Cardioid - Shure SM-57, Neumann TLM103

Hypercardioid - Beyer M201, AT25

Figure 8 (Bidirectional) RCA 44BX

Condeser mics which use two back to back diaphragms can simulate several patterns by electronically combining the two diaphragms in different configurations (e.g., combining a figure 8 pattern and and an omni pattern results in a cardioid pattern).

Chris :

I think I get it. But then, how does an omnidirectional condensor work? And what is a common type of pressure mic that I can think of to try to visualize the difference between a pressure transducer and a pressure gradient mic? If I'm understanding you so far, most of the mics used in the recording industry are all pressure gradient mics, right?

Anybody else still following this thread, or have I become a class of one? Speak up guys! I don't want to be the only one asking dumb questions around here.
Whoops, you answered my last question while I was posting. Thanks.

Kelly Holdridge :

I'm especially interested in how B&K made a microphone with flat response. I'd imagine it's a mix of various patterns and electrical tricks... Also, how much did it cost and was it a big deal when they announced it?

Harvey Gerst :

Brüel & Kjćr has been making measurement mics for about 50 years now. They're omnidirectional, and flat (within a fer tenths of a dB) from about 5Hz to 40kHz, although the have some models that are flat down to 1Hz and other models that are flat to about 140kHz. Every mic manufacturers uses B&K to see what their own mics are doing; the test is very simple:

You point the B&K mic and the mic you wanna test at any sound source and record the two response curves. You subtract the B&K results from the mic under test, and any differences from the B&K response - well, that's the your mic's response curve.

They made two basic types of omni test mics - one for pressure field (on axis) and one for diffuse field (90 degrees off axis). Their DPA web site (DPA is their name for their studio type mics) contains a whole bunch of good, objective info about the differences between small and large diaphragm mics, but it's pretty techie oriented. (Bottom line: small diaphragm mics have higher noise, flatter response, greater dynamic range, large diaphragm mics have higher output, lower noise, and less dynamic range and frequency response.)

Dick Rosmini in California was a big champion of using B&K test mics for recording and we usta have long arguments about it, since I found them kinda boring.

Tom Cram :

I had a love affair with using Earthworks omnis for a while, then I got over it. I still love them for recording acoustic guitar, overheads and sometimes as ambient mics. But the IDEA of very flat, full freq omni is cooler than the reality. If I was doing more classical or location sessions instead of Rawk sessions, maybe I'd use them more. Flat is not necessarily the best IMHO.

Chris F :

The best recorded piano sound I've ever heard was on a couple of Lynne Arriale trio records from the 90's, and was recorded using a pair of Earthworks omni's.... I don't know the model number, but they were tapered down to about pencil diameter at the business end. Fantastic sound, but I don't know how they processed it. The beauty of the sound also had a lot to do with her playing, which is stellar, especially as regards tone.

I have heard that those mics do not color the sound in any way. This seems both intriguing and frightening at the same time, since crystal clear sound exposes all of the warts on both instrument and player. My Baldwin grand is a decent piano, but it has plenty of warts (the doctor wants about $3000 to remove them, and I've already spent about $1500). If I understand the meaning of "omnidirectional" correctly, it means that when you use them you had better use them with the intention of picking up every sound in the room, which means either a very controlled or very forgiving environment would be needed for best results. How many home reccer's can afford THAT luxury?

On the other hand, that piano sound is the stuff that dreams are made of....

Harvey Gerst :

Let's review what we've learned so far.Okay, no pop quiz today, but let's review some of the stuff we've gone thru so far.

For most applications, the 3 basic mic designs are:

1. Condenser mics
2. Dynamic (moving coil)
3. Ribbon mics (a special class of dynamic mics)

The basic Polar patterns are:

1. Omni- directional (pressure)
2. Uni-directional (cardioid)
3. Bi-directional (figure 8)
4. Hyper-cardioid

True omni-directional mics have a sealed back chamber and only allow sound to hit the front of the diaphragm. The other polar patterns are created by using "pressure gradient" techniques to delay and let some of the sound hit the back of the diaphragm.

Condenser mics can be made in small (1/2" or smaller), medium (5/8" to 7/8"), and large diaphragm (!" and larger) sizes. Small diaphragm condensor mics have these advantages:

1. Flatter, extended frequency response
2. Higher spl levels
3. Better off-axis response
4. Greater accuracy

They have these disadvantages:

1. Lower output levels
2. Higher self-noise

Large diaphragm condensor mics have these advantages:

1. Higher output levels
2. Lower self-noise

They have these disadvantages:

1. Poorer limited frequency response
2. Lower spl levels
3. Uneven off-axis response
4. Less accuracy

However, some of the resonances in a large diaphragm condensor mic can be very pleasing and musical, and can often compliment the voice and some instruments very well.

"Pure" pressure mics do not have proximity effect (bass buildup as you get closer) - all pressure gradient mics DO have proximity effect (dual diaphragm condensor omnis have the least, then cardioids, then hypercardioids, then figure 8, which has the most proximity effect).

You would use small diaphragm condensor omnis where you want the greatest accuracy or in high level situations where self-noise isn't a factor. Large diaphragm condensor mics are better used for quiet sources, or where you want a particular type of complimentary coloration.

We'll get into choice of mic, patterns and placements in the next installment.

Kelly Holdridge :

I we've been using our EV-235a as our harmonica pickup; it was "the leftover mic." The lack of proximity effect changes with this application, because Todd's hands are cupped around it. Together with an' old crappy solid-state Crate w/ lots of chorus and reverb (and gain), you can get a KILLER harmonica tone. Then it's a '57 into the board...

One more thing: I've noticed that cheap electronics often incorporate a condensor mic (remember those old tape recorders?). I figured for the price, they'd just put in dynamics... Are these real condensor-type mics?

Harvey Gerst :

Yes, they're small, cheap electret (pre-polarized) mics, usually available from Taiwan for around 15 or 20 cents each. Much easier to mass produce than a dynamic mic.

The Axis :

Just another perspective on the flat omnis:

Many people want something that will make their voice or instrument sound "better" than it really sounds ...add "excitement", etc. The flat omnis DON'T do that.

BUT, if you have an instrument that DOES already sound good, the flat-omnis are much better at *capturing* that sound.

So I have come to recognizing the widely varying opinions regarding these as simply representing two different philosophies of recording. Some want the recording chain to be part of the "tone enhancement" system. Others want a transparent "capture system". I tried a lot of "highly recommended" mikes that sounded just horrible to me until I found and settled in with the E-Wks QTC1s.

If you are one of the people who keeps wondering why it is so hard to capture a good sound from an instrument you are happy with, they are definitely worth a try.

Harvey Gerst :

A few more things to think about

OK, before we get into what mic to use for what purpose, and where to place it, here are a few more things you hafta be aware of. One is called "musical intrument radiation patterns" and the other is "near-field placement vs. far-field placement".

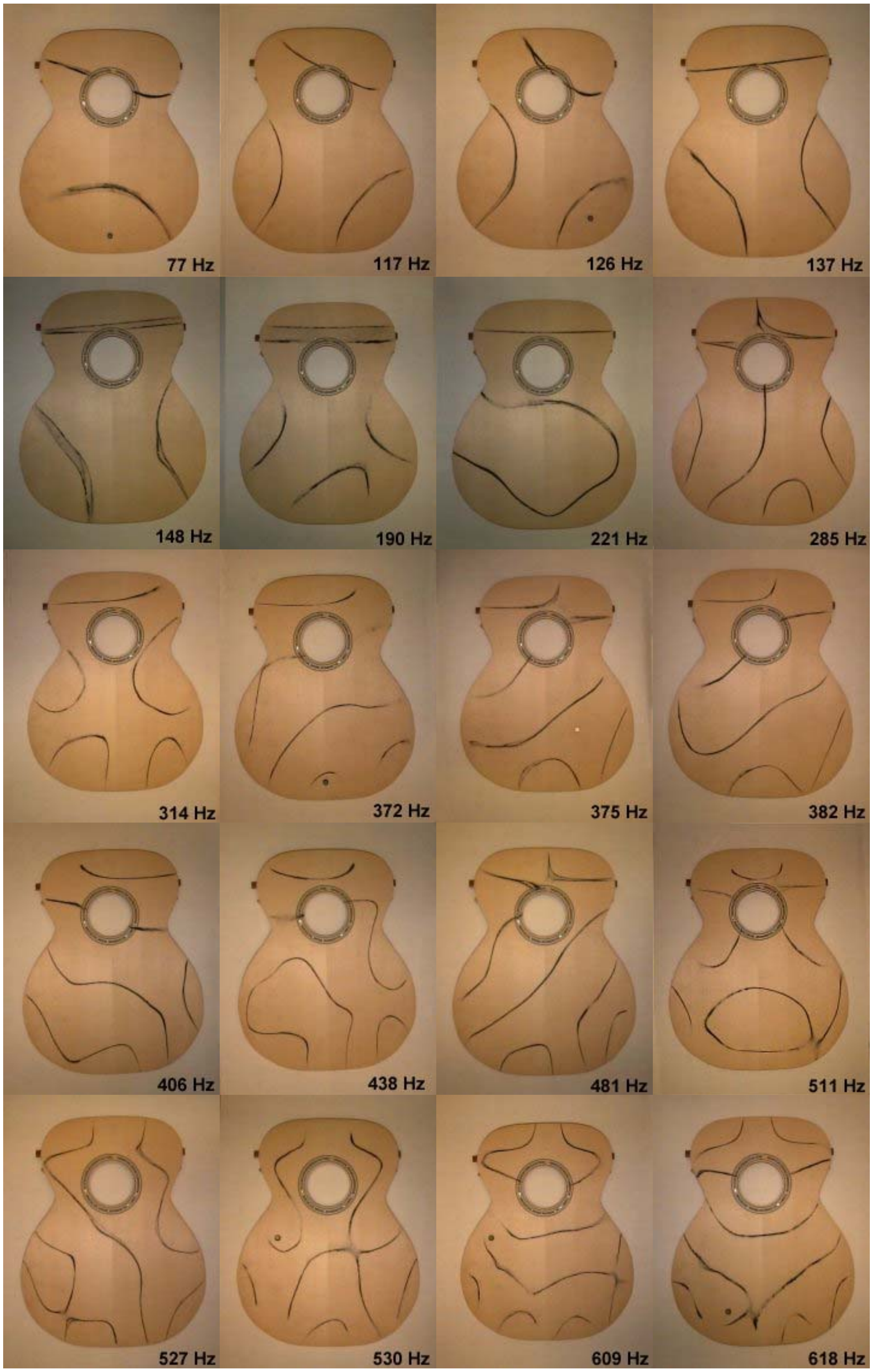
The most common question here is "how do I mic an acoustic guitar?", followed by vocal mic techniques. Let's look at the first question because it's more complex than it appears and it's actually made up of two parts.

Musical intrument radiation patterns

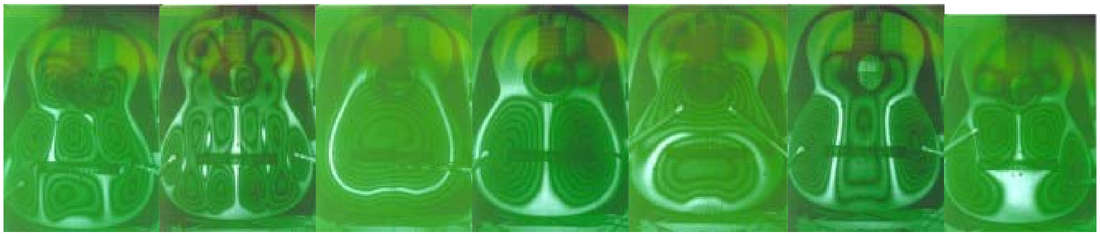
Guitars, violins, stringed instruments, in fact, all instruments radiate notes differently at different frequencies!! Read that again: Guitars, violins, stringed instruments, in fact, all instruments radiate notes differently at different frequencies!!

What does that mean exactly? It means that different parts of the instrument's body are used to produce different notes! Just pointing a mic at a guitar is no guarantee that you'll get what you want. Unless you understand how guitars generate sound, the best you can hope for is to somehow get lucky. Here are two links that show how the guitar top changes with each note:

Chladni guitar top radiation patterns http://www.phys.unsw.edu.au/music/guitar/guitarchladni_engl.html



Radiation patterns http://www.astro.cf.ac.uk/groups/acoucomp/MAGResMeth.html#fotos3



As you can see, different notes come from different places on a guitar, which brings us to the next section:

Near-field placement vs. far-field placement

Ok, so what the hell does that mean? Well, let's do a thought experiment to illustrate this concept:

Think of a tall column of speakers - about 6 feet tall, with woofers on the bottom, midrange speakers in the middle, and tweeters at the top. Now imagine that you walk right up to it and put your ear about 4" away from the system; what will you hear?

If you answered that it depends on whether your ear is near the tweeters, mids, or woofers, you're absolutely right. So where would you hafta stand to hear the whole system evenly balanced? At least 6 feet away is the correct answer - and that 6' away point is the boundry between the "near-field" and the "far-field" in this example. Any closer than 6 feet and you don't hear the whole sound, because you're in the "near field".

Now let's look at a typical acoustic guitar. The body is about 2 feet across. Put a mic any closer than 2 feet and you're in the "near field" of the guitar, and those two links I posted show you that you will be hearing uneven sound, depending on the note being played.

So, the first rule to remember is: "The near field distance is defined as being equal to the length of the longest part of the vibrating section of the istrument."

The second rule to remember is: "Inside the near field of an instrument, the sound will change drastically with different mic placements".

We'll get into mic choices, polar patterns, and mic placements in our next installment, but this "radiation pattern" and "near-field" vs. "far-field" stuff is really important to remember when you're trying to get a good instrumental sound.

Chris F :

This makes perfect sense, but it also brings up a couple of points for later if I'm understanding you right:

a) the person who is **playing** the instrument is in the near field, and if they're an accomplished musician it means that they've been practicing for years and years in the near field, which would mean that there's at least a decent chance that the person creating the music doesn't really know what the instrument really sounds like to someone else when they're playing it. And a microphone is the proverbial **someone else** in this situation. This might explain why many acoustic musicians become so confused/disconcerted when asked to wear phones in the studio....because they are accustomed to hearing only from a certain place in the near field and reacting to that, and all of a sudden someone has moved their "ears" to another location by making them wear headphones. Either that, or I'm making this sh*t up as a rationalization for why I hate recording with phones on.... it makes me feel as if I'm suddenly not playing the same instrument anymore.

b) If you get further away than 6 feet, you might be hearing "the whole sound", but you're also hearing more than that, because the further you move away from the sound source, the more your listening environment (i.e. - room or hall) is coloring your sound with reflections of some sort. So this is why the speakers mounted on either side of the desk in most studios are called "near field monitors", right?

A couple of questions related to (a) above:
c) For stringed instruments of the fretless variety, is the sound of the string vibrating on the fingerboard (i.e. - the "growl" of the low notes on a Double Bass) also part of this "near field distance", or do you only count the length of the body itself? I'm only asking because this would change the length of "near field" somewhat.

d) For a grand piano, do you count the length of the longest (bass) portion of the soundboard? And how do you decide whether to mic the top or bottom of it?

(Harvey)*We'll get into mic choices, polar patterns, and mic placements in our next installment, but this "radiation pattern" and "near-field" vs. "far-field" stuff is really important to remember when you're trying to get a good instrumental sound.*

e) So any time you mic in the near field, you're really getting an incomplete sound, and if you use only one mic, or two or more mics placed more that about 5" apart, you're recording an "artificial" or "manufactured" sound since your ears could never pick up that sound in a natural acoustic setting. How would the "polar pattern" of a set of human ears be described using general microphone terminology? Stereo omnidirectional?

Harvey Gerst:

a) That's probably a part of it, and it also explains why a lot of guitarists like my over the shoulder technique of guitar miking.

b) Yes, as you move further back, more of the room comes into play. That's also when you start changing polar patterns to compensate, or to use more of the room sound (but that's all covered in the next installment). With regards to monitors, yes they are in "your" nearfield so you hear them before you hear any room reflections.

c) Since the body accounts for the bulk of the instrument's radiating energy, I tend to just consider the widest part of the body itself, unless you're close-miking.

d) Piano is one of the hardest instruments to record and I planned to go into that in greater detail later. For now, I consider the widest dimension as the length, and I mic from out front if it's a solo instrument, or in close if it's to sit in a mix.

e) Yes, but that "artificial" or "manufactured" sound is not always bad, if it works better in the mix. Ears are basically pressure transducers with increased directionality at higher frequencies.

c7sus :

What I wanna know is why the field of the instrument is the width (length?) of the body and not the length of the string..... which as you play changes constantly.....

Now I may be getting way ahead of myself..... but if the field is 2 feet out from the front of my guitar, wouldn't I want to use the tightest pattern possible to capture just the sound of the instrument and not an omni that is gonna pick up the room..... then, since I'm 2 feet from my source, won't I start running into S/N problems......

Harvey Gerst :

Ahhh, the light is coming on. Yes, when you're out of the nearfield, you start thinking about other mic polar patterns. There's where you might wanna switch to larger diaphragm mics (remember condenser diaphragm size determines ouput level, all else being equal). But you give up some accuracy when you trade off to a larger size.

The Axis :

Near Field...

(to C7sus:)

There is no sharp dividing line between the "near" and "far" fields. The differences just gradually fade into obscurity. That is why it is somewhat irrelevent what the length of the string is. The entire guitar body is always vibrating, including neck, headstock, strings, and body top, sides, and back..

My experience on acoustic guitar is that omnis sound more natural because they are more similar to the human ear, which is *closer* to omni than cardioid. They are also generally flatter (more accurate) in frequency response. If you move your head around in front of someone playing guitar, the tone does change somewhat, but it is not the same dramatic differences as when you move a cardioid mike around slightly (wild variations in tone).

Even well inside the "near field" (4 - 8 inches) an omni sounds much more natural than a cardioid. Then the S/N ratio problem is solved because the guitar is so much *relatively* louder than any other noise in the room. Having said that though, I keep my mikes about 14-18" out, because it does allow the sound to come together a little better.

An experiment is a very good thing on this phenomenon. Many people (myself included) have been virtually brainwashed into thinking that cardioid is the only way to go. A brief experiment with a good omni will blow your mind !

Chris Shaeffer :

Another concideration after looking at the resonance maps...

I would imagine that different instruments resonate differently- obviously- and we can easily here the different timbres of different accoustic instruments. Different guitars can even sound vastly different.

Question is do similar instruments project the same frequencies from the same areas? For example, do ALL guitars tend to sound basey in the same area?

Seems like this would be handy stuff to know when deciding where to start with near-field miking. I can also see that the different mike patterns will interact with different areas of sound projecting from the instrument...

I'm imagining a figure-8 pattern mike placed right between a guitar and a reflective wall that are 4 feet apart with the guitar facing the wall.

This is really neat stuff to think about! I'm looking forward to applying it. (and seeing how well/poorly those ideas work!)

Harvey Gerst :

Chapter 8 - Fortress Of Doom

Ok, one more quick review and we're off into placement and mic choice land:

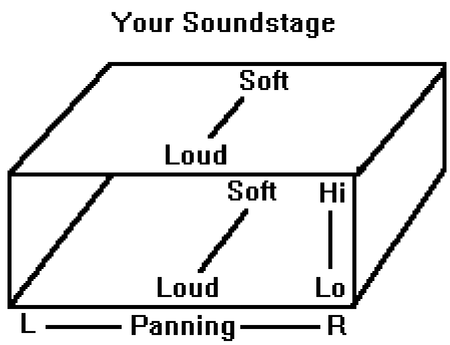
1. Small mics generally tend to be more accurate than large mics. Large mics are generally more flattering than small mics.
2. Omni mics generally have the greatest accuracy but the smaller most accurate omnis have a higher self-noise level, but they can handle higher SPLs as well.
3. Pressure gradient mics (cardioid, hyper-cardioid, and figure 8) use delayed sound coming into the backside of the diaphragm to create their patterns.

Any mic can be used to record any sound

Think about that for a minute. A mic doesn't care what it's recording, but some mics are better suited for certain things. If you don't have the one "perfect" mic for the job, any mic will work, with some trade-offs in sound quality. Understanding what those "trade-offs" are is the hard part.

How does this part fit into the song?

Does it need to be miked in stereo or will mono work? Is it gonna be an upfront part or does it hafta blend in? Is there another instrument operating in the same frequency range that might cause a conflict? The right choice of mics can help in all these situations, but you hafta make sure BEFORE you set up any mics just exactly what it is that you hope to accomplish. Here is how I think about where a part fits in:



Notice there are 3 important elements to this box:

1. Panning (Left to Right) - Very useful for separating instruments that occupy the same frequency range.

2. Level (Front to Back) - In combination with reverb, this creates the illusion of near and far, and can also seperate instruments in the same range.

3. Frequency (Lo to Hi) - the most overlooked aspect of getting a good blend when you're first starting out. Most people solo a track and then work to get a killer sound (bass, guitar, whatever), then move on to the next track. Wrong way to think. Instead, think about the song; which instrument should cover the bottom octave, electric bass or kick drum? If it's the kick, roll off some of the bottom on the electric bass and listen to make sure the two instruments aren't fighting for the same space.

Is the vocal important? Put it in the center, right up front. Are the guitars conflicting with the vocal? Move them out of the way with the pan control.

But what if it's just a solo guitar or piano track? Ahhh, there's where you need to decide if a stereo recording would be best. If it's an acompaniment to a vocal, a stereo recorded guitar or piano can sound very nice contrasted with a mono vocal.

Sorry this has turned into a rambling diatribe, but these are things that people tend to overlook in their haste to record stuff. But it's exactly this stuff that determines what mic, polar pattern, and placement you should be using - before you even plug in the first mic. We'll cover exactly that part next - I promise.

Chris F:

The one part I'm not sure I understand is the "near-far" or "depth" part - is this aspect actually created by micing certain layers closer in the "near field" than others, or is this some kind of illusion produced by mechanical means?

Harvey Gerst :

Near-far "depth" is achieved by using a combination of close and distant miking techniques AND the judicious use of reverb to place instruments at different distances in the mix.

For example, you may wanna record the vocals at "point blank" range and add just a touch of a catherdal reverb, so that the vocal is still up close, but you get a sense of a larger room. Strings might require that you mic from further away, roll off a little bit of the top end and add more reverb to simulate how they would sound if they were in a large room.

You control depth of field by careful use of mic placement, final mx level, and reverb when planning the mix. Just remember that heavy reverb tends to blur the sound of the instrument, so don't overdo. I know of one engineer who worked for two weeks just on the fine tuning of the eq on the reverb for the snare.

Chris Shaeffer :

(on Harvey's) *1. Small mics generally tend to be more accurate than large mics. Large mics are generally more flattering than small mics.*

Now that I see WHY this is true (smaller diaphrams move more quickly in response to the details of a sound) it gets me thinking.

1) "Accurate" doesn't always mean "Sounds better" and "Flattering" doesn't mean "Accurate." I know that seems like a 'duh!' statement but it seems important.

2) I wonder what it is about human hearing that makes certain kinds of "less accurate" sound better to us. It reminds me of the difference between consumer stereo speakers (flattering) and monitors (accurate.)

3) I also notice that there doesn't seem to be any measurement that quantifies the accuracy or speed of responsiveness of a mike- frequency response isn't really the same thing. Also, no measurement that I know of for that elusive term "color." Is there more to the "color" of a mike (or amp, or, speaker, or...) than the frequency response?

and finally....

4) 1-3 above seem like clear proof to me that choosing a mike and placement that sounds the "best" for a given source is going to involve knowledge of mike and sound source characteristics (which we seem to have covered *really* well), some ideas of where to start with placing those mikes, creativity, perseverance, and luck. I suppose that experience can make up for the lack of luck.

It makes me want to take REALLY good notes about how I get the sounds I like- as well as the sounds I don't like. This seems like an art and science that deserves a lot of attention. I'm beginning to see every mike/sound source/placement experiement as time well spent even if it fails. Good stuff to have it one's head.

Harvey Gerst :

Some great insights, Chris (and from the rest of you as well). Just hang in there gang, we're almost thru. This last part will be a series of multiple posts, since we'll be covering so many techniques and instruments (in as much detail as possible).

Manchild :

Is it true that mic frequency plays apart of the source that you are recording? and that you are much better off using a mic that is as close to the frequency of the soruce you are trying to record? ie. piano, singing voice, drums, guitar, flute,etc,etc.....

Harvey Gerst :

Yes, and no!! The mic's frequency characteristics are of course a vital part of deciding which mic to use where, but it's often a choice of complimenting instead of capturing. In other words, sometimes you use a mic to flatter and enhance the sound, not because it has a similar range or is the most accurate choice.

PART II
(microphone placement techniques)

Harvey Gerst :

The guitar - Mic choices, patterns, and placements

Ok, here we go, pinched sciatic nerve be damned!!

Miking an acoustic guitar

Many acoustic guitars today have built in pickups, and it's gonna hafta be your choice whether you add that to the mix or not - that's a whole 'nother subject. Before you reach for a mic, you hafta decide a few things:

Is it a solo guitar, strictly as a backdrop for vocal, or is it one part of a group mix (where there'll be other instruments like drums and bass and electric guitars going on)? Does it need to be recorded in stereo or is mono ok? Is it gonna wind up being in your face, or buried in the mix?

Solo Guitar or Background For Vocal?

If it's a great sounding guitar, and you have a good room, you want to use the best mics you have and record in stereo. You can use omnis, or a pair of good cardioids in an X/Y configuration (capsules almost touching, angle of about 110 degrees between the two mics) and about two feet out from the instrument.

A dynamic or condenser mic will work fine as long as the mic has a fairly smooth response. Smaller condenser mics are usually more accurate, but if it's not a killer instrument, don't be afraid to try large diaphragm mics to get a more flattering sound. The mics should be pretty closly matched otherwise the stereo image can shift as you play different notes.

If the sound ain't working for you, that's the time to move in closer and see if you can find spots nearer the guiitar that produce a better tone (even if it's just for that song). Try to get as close as possible to the final sound you want BEFORE you reach for eq and/or effects.

After you get the tone damn near perfect from placement and selection, then do a little touchup with the eq to nail it. (If you hafta boost or cut more than 4 dB in any frequency range, you either haven't got the placement right yet, or it's a really crappy guitar.)

Acoustic Guitar as a Rock Track With a Band

You need a tone that's gonna cut thru the other instruments and if there's gonna be drums, bass, electric guitars going at the same time, record the guitar on the thin side (some bass cut and treble boost). Make it brighter than you normally like it, and don't worry about how it sounds soloed - it's how it sounds when it's all mixed that will count. I usually mic in close (about 6 to 8"), from slightly below, looking up directly at the bridge. Roll off the bass below 100 Hz, and boost around 2 to 4 kHz (move the frequency around to where it sounds bright, but not shrill).

The Singer/Songwriter Syndrome

The singer also want to play guitar at the same time, and you want some decent separations between the vocals and the guitar. One trick is to use a X/Y stereo pair of small cardioids down low, aimed at the guitar, hile you position a large diaphragm mic at the singers forehead, tilted just slightly forward, toward his/her nose.

Some Points To Ponder

These techniques should work for banjo, mandolin, 12 string, uke, and other small stringed instruments. But sometimes they don't always work as planned. If you're not hearing the sound you want, try moving the mic around, even to the point of miking the side of the instrument instead of the front. Violins, cellos, and upright basses are a whole special category which will be discussed later.

A good trick is to stick your finger in one ear and move around till you find a spot that sounds good, then put the mic there for starters. Remember that each guitar is different, each mic is different, each room is different, and sometimes just going up or down a 1/2 step will change everything. Starting from the outside edge of the "nearfield" is a great starting point.

Some Mics To Try First:

Dynamics: Shure SM57 - Sennheiser 421 - Beyer M201

Ribbons: Beyers, RCA, any ribbon mic.

Small Condensers: Oktava MC012, Marshall 603S, AT 4041, Neumann KM184, any small cardioid or omni condenser mic.

Large Condensers: These mics add a great deal of color to the sound, so "try" anything you happen to own. It may work great or shitty - you never know.

End of the first part - more to come!

KaBudokan :

Harvey, what kind of placement strategies (or starting points) would you suggest for mixing large and small diaphraghm condensor mics for micing acoustic guitar?

I have an NT2 that I've been using to mic my acoustic with mixed results. I picked up one of the Marshall MXL1000's from ebay the other day ($50, I figured even a poor grad student like me couldn't go wrong), and I am wondering what may be the best approach if I am going for a stereo mix using those two mics.

I'll experiment, of course, just wondergin what good starting points may be. I'm thinking try to find a solid sound I am happy with using the MXL, and then compliment with the NT2... (Knowing the NT2 is a bit harsh on guitar, I'll probably try to find a spot which mellows that shrillness.)

Harvey Gerst

Start around 2 feet out with the two mics set up in an X/Y array and see if you luck out. If not, try the Marshall 1000 by itself over your shoulder, figure out what's missing and then try to get that from NT2. Take the ball off the 1000 and put it in a closet somewhere.

I think the 1000 should be the main sound for your guitar with the NT2 as kind of a fill-in mic. Read my whole description of guitar miking techniques a few posts above this one - I tried to get pretty detailed.

KaBudokan :

I like the idea of the "over the shoulder technique." I am hoping between the two mic's I will be able to get a good, interesting sound. I can always pick up another Marshall to do stereo micing with too.

Chris F:

a) The mic placement ideas for acoustic guitar make a lot of sense. I guess the part I didn't realize was the "wild-card" factor of the large diaphragm mics. I understand that they "color" the sound, but from what you seem to be saying, there's no way to predict in just what way they will color it....which is kind of a bummer. I notice when people are using them for voice that there is a certain "depth" added to the spoken sound that the smaller condensers don't seem to add, but I have no idea how this would translate to instruments.

b) I'm happy to wait for the installment on grand piano, but even with my own NEWBIE mic placement I notice that the mxl 603s gets a much better sound than anything I've used before (including AT Pro 37). The "xy" thing is going to require a special mounting device, and it probably means I need to order at least 1 or 2 new mics for the other instruments since I want to do 90% live recording.

c) What about acoustic bass? The biggest problem I have noticed with trying to record this instrument is that there is one note (the open "D") string which seems to record way hotter than the rest of the range of the instrument, so that when the rest of the range is in balance and sounds great, that D comes in and bottoms the whole track out. How should I deal with this? Bass roll of on the board while recording? Moving the mics further away? This may be an EQ question rather than a mic question...

d) I know a lot depends on the instrument, the player, and the mic, BUT....when recording acoustic bass, do you often use a large diaphragm mic? If so, have you ever used the V67G for this purpose? Over at my bass site, there is a lot of heated discussion about the issue, but it's coming from players - most of whom (like myself) don't know much about recording - rather than engineers. I'd love to hear your input, even though I realize that there is no definitive answer for all situations.

Harvey Gerst :

a) It's a major problem, Chris, especially when talking about this new crop of low cost, condenser microphones from Russia and China. The Quality Control from many of the distributors leaves a lot to be desired. Some capsules have a very ragged and peaky top end, while others either have a diminished bottom end or they're bloated in close. The depth you hear on vocals is usually from the "proximity effect" when singing in the microphone's near field, boosting the bass at around 250 to 400 Hz to produce a fullness that's usually very desirable.

b) Yes, the 603S mics are an amazing buy right now for x/y use and many other applications. Acoustic bass, cello, flute, and fiddle are other uses for the 603S, which I'll try to cover in further installments.

c) There are several mic positions that can lessen that "hyped D" effect, like miking on the bass side of the bridge, and even with a mic stuffed part way inside the "f" hole. You can also use a parametric equalizer to lessen the effect. I'll get into all that in another posting later in the series.

d) There are times when a large diaphragm mic can sound very good on upright bass, but it's usually from the middle of the nearfield (about 2 feet)to the far edge of the nearfield (about 42")where it sounds best. Any closer and the proximity effect will produce a boomy upper bass tone that destroys any definition and detail in the tone.

Dobro:

"*The Singer/Songwriter Syndrome*

The singer also want to play guitar at the same time, and you want some decent separations between the vocals and the guitar. One trick is to use a X/Y stereo pair of small cardioids down low..."

How low? Below the guitar and pointing up? Under the guitar basically? This is the first time I've heard about this. How far out? Doesn't it have to be close in to maximize the guitar and reduce the vocal? I'm gonna find out eventually, but you can help the process along if you like.

Harvey Garst :

As usual, it will depend on the guitar, but right near the bottom of the lower bout is a good place to start. Getting in close WILL reduce the vocal, but, as you now know, that can create other problems. Start at about 18" away with the mics slightly below and aiming JUST SLIGHTLY up toward the body. The trick is to get the vocal mic up high and close to the singer without using a windscreen.

c7sus

I have read about guys using music stands to isolate the instrument mics from the vocal mic in these situations. But I wonder if the added baffle of the music stand doesn't start causing it's own problems in this.

Harvey Garst :

That can sometimes create more problems that it will fix. In essence, you're creating a mechanical comb filter. Adding some padding can oftem minimize the combing effect, but you can also position the mics close to the stand and create a Blumlien pair which will give you very good separation. You can build a Blumlien stand by using a piece of plastic about 12"x 12" and gluing a mouse pad on each side.

Rjbutchko :

...as far as acoustic guitar is concerned, as long as that's the example we're working with, would it be a good idea to mic the guitar from a couple of feet away, as you suggest, and add a near-feild mic focusing on the freq's you wish to enhance in the mix? Provided, of course, that the whole mess goes mono. (?)

Harvey Garst :

As long as you observe the 3:1 rule (the second mic must be at least 3 times further away than the first mic, it might work well. It depends on moving that mic in the near field till you find the perfect spot.

Chris F :

You mentioned the "Hyped D" effect regarding acoustic bass in your last response. Did you do that only because I mentioned it first, or is the open D a problem on many acoustic basses?

I'm asking because both my carved bass and my plywood do this when I record, though neither does it at the fingered unison, octave, or double octave. My carved bass used to pop out a pretty ugly open "G" as well, but I tempered it by putting in some velvet padding at the nut on that string. Maybe I should try this for the D string as well, even though the "hyped" effect isn't all that noticable to the human ear when heard live. Hmmm....

Harvey Gerst :

Most basses, upright and electric usually have one, sometimes two, predominent strings. The old Fenders were very notable in that the bottom two strings sounded completely different from the two high strings. With upright basses, it's also a function of the resonant chamber, the porting, and the woods used. It's usually the two upper strings that create the most problems for recording.

You're in the near field when you're playing, so it may not be accurate, plus the G and D may be really booming off the upper bout.

Dobro :

Okay, I've been trying miking the gitbox from down under (no, I haven't booked a studio in Melbourne), and I'm getting good results. Close works better than 18-24" out, probably because of the room, I think.

I tried the 'forehead level, point it at your nose' approach for the vocal mic too, and that's good too, but because of the room I'm in, I actually get a slightly better sound by having the mic at nose level and pointing it at my forehead!

Harvey Gerst :

An interesting, easy experiment:

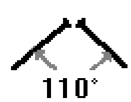
Put your hand out about one hand's width from your face, even with your mouth, and try to blow straight ahead. Feel where the air blast is actually hitting your hand. Surprise!!!!

If you're like most people, the blast will actually hit the bottom two fingers of your hand. We all tend to blow slightly downward, so putting a vocal mic at nose height (or higher) actually misses the bulk of the air blast that causes popping on words that have a "p", "f", "b", or "v" in them.

Tell the singer NOT to aim into the mic - just have them sing straight ahead, or set up another mic in the standard vocal mic position (the "dummy mic" trick), and have them sing into that one. Or put the lyrics in the standard vocal mic position, so that they hafta keep facing straight ahead to read the words.

X/Y miking setup

For some of you that may not know what "X/Y miking" is, here's a diagram of two cardioids set up for X/Y miking:



Notice the capsules are almost touching and the angle between them is 110 degrees. This can also work with two omni mics, but with lower stereo separation.

More on all this stuff later.

h kuhn :

Harvey, in the book from the schoeps-site they mention that an xy-pair should be angled 90° (also visible in the williams-diagram). What would be the advantage of the wider angle you recomend?

Tekker:

My guess would be for a wider stereo image and better separation......

Harvey Gerst :

Tekker is right. You use the angle to adjust the level of the center signal and the amount of left/right seperation. And for any angle greater than 90ş. you usually place the capsules one above the other, at their centerlines.

We'll be going deeper into stereo pair miking, talking about coincedence, near coincidence, and wide spaced mics - the 3 possible basic stereo methods. These 3 basic methods cover the Jecklin Disc, X/Y, Dual Figure 8s, Decca Tree, Blumlein Pair, ORTF, M/S, and wide spaced omnis, and cardioids.

Explaining (and visualizations) of each type will make it easier to understand how a method works and when to use it.

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c7sus :

Hey Harvey..... if it's not too much trouble I'd like to hear your opinion about the old FRAP stuff. I've got two... an old Model T and a custom one-of-kind that Arnie put together for me in 85-86.... he wasn't doing real well back then. You wouldn't happen to know if he's still around, would you? Just curious. I've always liked the sound of those FRAPs. The custom one has a ten position bass rolloff, phase switch, and hi and low Z outs, plus an electric jack as well. I mounted the pickup in each of my guitars myself because I couldn't find anyone that would take the time to do it right. But I managed to find the sweet spot on both my Martins and I've never heard any piezos that can hold a candle to them.

Harvey Gerst :

Sorry, I can't help you on this one. I got burned out on the music business and got out of it in 1978. They tracked me down and dragged me back into the business in 1987, so there's 9 years that are just a blank. I remember the FRAP name, but not much about it.

h kuhn :

Harvey, you recorded Ornette Coleman??? I just checked some of your old threads on the gm forum. Do you remember what year it was? Is the record still available? And, most important, the question is burning on my nails: Did you use individual mics for each musician or did you put up room mics (or a combination)?

To the "110° issue": wouldn't at this angle something that is actually behind the mics appear far left and right in the stereo image and thus corrupt the fidelity of the recording (suposing cardioid mics)?

Harvey Gerst :

Yes, I recorded Ornette when he was just starting out He played at the Saturday night jam session at "Georgia's Place" in El Monte, and I would pick him up and drive him there (with his white plastic saxophone). The house band consisted of Charlie Haden on bass, Freddie Gruber on drums, and I can't remember the piano player's name, but he was killer as well. It was in the early 60s sometime, as I recall, but I can't remember what year. I used a Berlant Concertone 20 tape deck and two Capps condensor mics (set about 6 feet apart).

The 110ş angle is just a suggestion. It seemed to give me a wider sound stage than the 90ş angle did and I prefer it. It really depends on what your recording, AND the room you're recording in.

Chris F :

I guess what I don't understand yet about the 110 degree concept is the difference between creating the stereo image with the mic placement as opposed to using the pan left/right on the mixer....this is a concept which had not even occured to me until now. I guess that's why it says "newbie" under my name....

Harvey Gerst :

You still pan the mics hard left and right, but the placement is critical for maintaining the accuracy of the stereo image. If done correctly, there are no phasing problems what so ever, and you get a perfect stereo spread. That's the goal of every two-mic stereo imaging system and they all have different advantages and disadvantages, depending on the source.

c7sus :

Ok, so I gave it a try.....

I got from the diagram that the mic bodies are 110 degrees to each other, so that puts the capsules at 70 degrees face to face.... right? The mics are an NTV and an NT2. I'm recording a 1985 Martin D-45 with light strings that are fairly worn.....

Anyway, I started at 2 feet with the faders set at unity and trim up about 3-4 dB for the NTV and up about 9-10 dB for the NT2. Bummer that the NTV is so much hotter...... all the EQ is set flat.

Anyway, the first pass I finally made was about 18" out with the mics above the soundhole angled down toward it. I got a fairly "strong" signal through but a lot of harshness which I am assuming is from all the gain on the Mackie pres.....

So I'm a bit disappointed right now..... so now what? The guitar sounds pretty "honky" too in the mids...... is it the mics or the placement or a combination of both? Hell, maybe that's just what the guitar sounds like.... I've got a VC6Q too, which allows a lot more gain without harshness..... but alas only one of those......

I'm thinking the next thing is close miking with the hotter mic behind the bridge and the NT2 at the 14th fret.....

h kuhn :

it is more common to use 2 identical mics for xy. the difference in frequency response between 2 different mics may cause phasing effects. BTW the nicest sound we got from an accoustic was with two mc012 about 70cm apart and 1m from the source, with omni capsules (this, of course, ist NOT coincident and has got nothing to do with xy!). Harveys over the shoulder tecnique was close second (but we had problems with noise from the player's skirt). The aim of the recording was to achieve a "being right there" feeling, lots of room while mono compatibility was no issue.

Wil Davis :

I notice that nobody has mentioned that much under-used (in my opinion) coincident set up of MS (Mid-Side). I never cease to be amazed at the results possible from what is essentially a very simple technique. The main advantages of using this method being (1) the two channels can be recorded and the stereo mixed at a later time, and changed as seen fit, and (2) the mono-compatibility of MS is just about perfect. Anyone else use it?

Elle :

Harvey, what would you use to record piano? (upright)

Harvey Gerst :

With an upright piano, there are no rules, but in general, I would use a couple of small condenser mics, but where is the big question, rather than what. Sound radiates from an upright piano from a lot of different directions. Opening the top and aiming a mic into the piano's innards usually results in very unpleasant jangle.

I'd first pull the piano awayfrom any nearby walls, and try a mic pointed at the back side of the piano (the sounding board), listen and decide what else was missing. Maybe a mic on the front somewhere as well, checking for phase, and trying to balance the two mics to get the full range sound without the jangle.

Every upright piano is different and you hafta experiment with different mic locations. Getting the mics further back in a good room will give you a more balanced sound, but room noise then becomes a factor.

Harvey Gerst :

Vocals - Why Are They So Hard To Get Right?

The two most asked questions are what's a good mic for acoustic guitar and what's a good mic for vocals. Most people actually want one mic for both, but if you've been following this whole thread, you know that the mic requirements for acoustic guitar are different than the mic requirements for vocals. So, what's so special about vocals and vocal mics?

There are three types of mics that are usually used for vocals:

1. Large condenser mics
2. Dynamic (moving coil) mics
3. Ribbon mics (a special class of dynamic mic design)

There are two types of patterns usually used for vocals:

1. Cardioid (most typical)

2. Bi-directional (Figure 8)

All of the above mics and patterns have "proximity effect" in common (more upper bass boost as you get closer to the mic).

With these mics, you can adjust the distance and the angle between the singer and the mic to get a wide variety of tonal effects till you find the right balance for a particular singer and song. Off axis response will often vary dramatically with large condenser and dynamic mics, and when coupled with the "proximity effect", you have a wide range of tones to choose from.

The general working range for most LD condenser and ribbon mics is anywhere from 6 to 18" away. Dynamic mics are usually best under 6" away. But there is no hard and fast rule there. For intimate softer ballads, you may want the singer to "eat the mic", recording them from 2" away, or even closer. Up close, wind blasts are a concern and a pop stopper, foam wind screen, or even both may be required.

Remember that "proximity effect" starts in the upper bass (around 400 Hz), and this is exactly the start of the human vocal range. It can add richness to a thinner voice, but as with most things, it can be overdone. You adjust "proximity effect" by adjusting the distance between the mic and the singer - closer for more, further back for less.

Use different mic angles to adjust the high frequency response - straight on for maximum highs, off axis for less highs.

As mentioned earlier, most singers breath blasts are aimed slightly downward, so try to get the mic above that blast when possible. I try to mic from about nose or forehead high, aimed slightly down towards the mouth, but if a person is more comfortable with a stage mic at mouth level, don't be afraid to give it a try.

Some condenser mics tend to have some bright high end peaks which may help a singer that doesn't have a lot of high frequency content in their voice, but it's all too easy to just end up with an overly bright vocal. You usually look for a mic with a smooth top end (like a ribbon), or a mic with a gentle high frequency rise.

With mics like the AKG C3000, some of the Rode mics, or the lower end LD ATs, watch for peaky high end response that may result in an overly bright vocal track that high end eq can't fix later.

Try to choose the mic that doesn't require any eq when recording, if possible. That's where the right sound begins. Use compression sparingly when doing the tracking - you can always add more later.

I try to avoid commiting to any effects while tracking, so that I have more options available during mixing. You can't turn the vocal reverb down later if you record with it during tracking.

Chris Fitzgerald has been doing some experimenting with upright bass mic placements and he might want to share some of the things he's found so far.

After that, we'll discuss miking drums, then grand pianos, then horns, and exotic instruments.

JerryD :

1. Would you ever mic vocals using two mics? To get a fuller sound? Is there another interesting technique for this?

2. What music in the public domain do you feel is well produced? ie miking, tones, ect...

Harvey Gerst :

1. If you mean would I mic a vocal in stereo, the answer is no, never. With most vocalists moving around even slightly, the phasing problems would be enormous. There are two times when I would use two or more mics on vocals:

   1. If the singer has a very wide dynamic range (from a whisper to shouting in the same song), I might use two mics recording to two tracks; one mic set so that it doesn't clip on the loud parts, and the second mic set up to pick up just the soft parts and to hell with clipping on the loud parts, then compile them in the mix, bringing up the soft mic in the mix during the soft sections, and killing it during the loud parts.

   2. An interesting technique that David Bowie used is to set up a second mic about 15 feet away and gate it so that it only comes on during louder parts, and then a third mic set up around 30 feet away and gated to only come on during the very loudest part of the vocals. This gives you two natural delays of 15ms and 30 ms, yet keeps the main vocals very up front. Very cool trick.

2. "Public Domain" has a very specific meaning in the music business. It means music that is free of copyright protection and may be recorded by anyone without having to pay royalties (like old folk songs). I assume you meant what music is out right now that I think is well produced etc.

   Anything produced by George Massenburg or Al Schmitt will always be well recorded and an inspiration for me to try and come close, even though I know I won't. There are some artists I don't even particularly like, but their recordings are beautifully recorded and well pr**oduced.**

h kuhn :

Harvey, what is the advantage/disadvantage of using a figuer 8 mic over a cardioid for single voice? Would you set up the figure of 8 differently (absorbing surface at the back?)?

Harvey Gerst :

I was primarily thinking of Figure 8 ribbon mics when I wrote that but some of it holds true for all figure 8 patterns. Figure 8 patterns have the most proximity effect possible, which can really enhance some vocals. Ribbons have a smooth, silky sound to them which compliments a great number of voices. Figure 8 patterns have the smoothest off-axis response of all gradient polar patterns.

Forgot to add that yes, I use some form of absorbtion for the back side when using figure 8 mics for vocals.

h kuhn :

Is it true that the "rudy van gelder sound" is mostly due to the use of ribbons?? As this is a sound I am absolutely looking for, what mics in the lower end could bring me a bit nearer (I certainly can't justify buying a RCA 77 or a Royer, bummer)?

Forgot to mention: I don't like the idea of a hypercardioid ribbon like the m260, because it would be nice to have the option of using it for m+s!

Harvey Gerst :

Ribbons, and a few Neumanns, I believe. Actually nobody knows Rudy's exact methods of recording and that is still a jealously guarded secret. He would often move the mics into weird positions purposely to confuse people who were curious. The Beyer M260 would make a good M mic, while the Beyer 160? (or is it the 130) would be a good S mic for MS recording.

And I forgot to mention that if you use the Beyer ribbon mics, get them modified by Stephen Sank in Albuqueque who re-ribbons them to where they sound like RCA 77DX mics for about $125 per mic. Try to find the Beyers used and beat-up and cheap, then send them to Stephen. Stephen's father, Jon, designed many of the original RCA ribbon mics.

chessparov :

re: large diaphram dynamics for vocals

Does anyone else besides me also think that large diaphram mikes like the Sennheiser 421 (I have one), Electro-Voice 520, and Shure SM-7 can work great for vocals?

a) I've read that Bonnie Raitt uses the EV 520 for her studio tracks, and that the Rolling Stones used an SM-7 for working on Voodoo Lounge (& other albums possibly). Since these microphones seem to be suited for blues/r&b rock styles I'm surprised not to see more interest at this bbs in them, especially for home recording where the lesser sensitivity to room acoustics can be a benefit.

b) A more appropriate question from me would be how your experience has been using large diaphram dynamic microphones for vocals, and how to use them more effectively for us home recordists. I notice a lot of people recommend using EQ to cut the midrange frequencies even after proper mike placement (2000 to 3000 hertz) about 2 or 3 decibels. Does that seem typical to you also?

Harvey Gerst :

a) The largest selling record of all time (75 million albums) used a Shure SM-7 for the main vocals instead of the artist's usual large condenser mic. The SM-7 is a pretty standard vocal mic in Nashnille, and it sits very well next to a Neumann U47 or a Telefunken 251 for some voices. The Sennheiser 421 was designed primarily as a vocal mic when it first came out, and it's still a great mic for many vocals.

b) No, it doesn't "seem typical" to me. If anything, I'll sometimes boost those frequencies slightly, then cut the same frequencies in other tracks to give the vocals more separation in a mix.

Keep in mind that it also depends on mic choice, since some mics do have an artificial boost (or peakyness) in the 2000 to 8000 Hz region. High, narrow high frequency peaks in microphone response drive me crazy, since they can't be tamed with general eq without dulling down the whole top end, and it takes forever to track them down using surgical parametric eqs.
As I mentioned many times in this thead, mic selection and placement are everything. Use distance to adjust your low frequency response, and use angle to adjust your high frequency response.

By the way, the record that sold 75 million copies and used a Shure SM-7 as the main vocal mic was.....

.....Michael Jackson's "Thriller".

Dolemite :

Harvey, don't you have that actual SM-7 used on Thriller? I think I remember reading that on r.a.p.

Harvey Gerst :

When I first got it, I thought it was the SM-7 that Bruce Swedien used on Michael, but I'm not sure any more. This is one of Bruce Swedien's SM-7 mics that he had around the time "Thriller" was cut. It still has his name on it in blue Dymo labeler tape that he puts on all his mics. When I bought it, I didn't know it originally belonged to Bruce Swedien - I bought it because it was a great mic.

The SM-7 stays on a stand in the studio 24 hours a day, wired up and ready to go. It's one of 5 mics that are ready to be used at all times (the other 4 are the Audix TR-40 omni, the Neumann TLM-103, the Marshall MXL-V67G, and a Shure SM-57).

The Audix TR-40 omni is used for misc. acoustic guitars, flute, violin, and various percussion instruments.

The Shure 57 is ready for adding electric guitar solos and overdubs.

The other 3 (the SM-7, the TLM-103, and the V67G) are first grabs for vocals, with the LOMO/MC012, and the ribbon mics standing by for other flavors. I'll also try these mics for horns, if a Sennheiser 421 doesn't give me what I want.

JerryD :

What is different about a valve microphone? How do these compare to a FET type mic? Is the rant about Rode mics, NTK and NT1000, mostly hype?

Harvey Gerst :

Well, lemme see if I can dispel some myths here. There are many reasons why a tube is used in some many condenser mics and is prefered to transistors (like FETs); some reasons make sense, while others are just hype and nonsense:

A condenser mic capsule requires seeing a high impedence source to keep the signal from being loaded down, usually in the multi-megohm to gigohm range. These can get pretty expensive. A tube, on the other hand, loves to see that kind of load, so it's a good match up.

Tubes, when used in a class A configuration, can be made to clip softly and symetrically, producing a more musical tone, rather than the sharp clipping characteristic of transistors.

Also, the heat (from the tube inside the mic housing) can serve as a built-in heater to drive humidity and moisture out of the case. All of the above traits are very good things. But there are some bad things as well.

You have to supple 12V DC heater and 250V plate voltages for the tube inside the mic, which require a separate power supply and a special cable running between the power supply and the mic. Tubes also tend to be a tad noisy. Different tubes can have different sonic properties which are either desirable, or not, depending on the manufacturer, or even the conditions on a particular manufacturing date.

All mics are a combination of hype and reality, mixed up and served by the manufacturer or the distributor, served up to fit their marketing plan. If you've been following this thread, you know there is no one microphone that will work best for everything. Large diaphragm vocal mics are the prime example - a mic might be perfect for one voice and sound like crap with another voice.

As far as tubes vs FET mic designs go, a good tube mic will beat the shit out of a bad FET mic, but a good FET mic will beat the shit out of a bad tube design. Tubes tend to smooth out the sound a little bit, hiding some of the things that might be harsh in other designs.

As far as the Rode mics are concerned, I haven't heard those two models. My main concern with the Rode mics is consistancy of sound from unit to unit.

h kuhn :

To my knowledge, the capsules of first series of rode mics (NT2, NT1) were manufactured in china. Today, the whole production takes place in australia. The consistency of the latest models seems to be very good.

Harvey, if you can get your hands on a NTK or an nt1000 to test it, that would be great! I think that all of us here would be very interested in your opinion.

Harvey Gerst :

I'm really glad to hear they got their consistancy problem solved. I'd like to give them another try. You may have notice that I've limited my comments to the earlier models that I did hear, but I expressed concern about the problem solely from their history.

I'll try to get to it, but things are getting crazy around here. I got a call from the head of Sound Projects microphones and he wants me to listen to his mics as well. I never planned to be the "poster boy" for low end mics and products, but in a way, it's kinda cool. I got burned a couple of times, listening to magazine reviews or reading the spec sheets, before I learned how to read between the lines.

I run a small recording studio in the middle of nowhere, and I'm on a tight budget, always looking for the best stuff at the lowest prices, the same as many of you. The only difference basically is that I've had a lotta years of listening to really good mics and understanding trade-offs in manufacturing. Since I'm not a "reviewer", but a potential "end user", I don't hafta worry about offending anybody, or trying to make a living from reviewing.

If some of my "impressions" (of products I've listened to) match your impressions, then we're probably on a similar wavelength, and my comments on other products may help you narrow some of your choices. I ain't no god; I'm just an old recording engineer, trying to get by, and help people avoid some of the mistakes I made when starting out.

Nearing the end.

Ok, here's some tips on miking some other instruments:

Horns, what mics and polar patterns to use

Most horns put out a lot of energy, so close miking is not a good idea. About 18" away is a good starting place. Start at the bell level and work your way up to pick up more breathtones and output from the keys, in the case of saxaphones.

Generally, you can use a large diaphragm dynamic, a ribbon, or a large diaphragm condenser mic with pretty good results. The Sennheiser 421 is a great choice for a dynamic, almost any kind of ribbon, or a !" condenser mic. Cardioid is usually the pattern of choice.

Choirs, pipe organs, and large stuff

Small diaphragm spaced condenser omnis or cardioids generally work best, pulled way back to capture some of the room as well. I usually split the sound source into 3rds and put a mic at the 1/3 and 2/3 points. (If the source is 60 feet wide, I'd start with mics 20 feet in and 20 feet apart and adjust from there.) How far away depends on how much reverb would sound good. I'd start with about 1/3 of the way back from the source, then move in closer or out further, depending on how much reverb I heard in the phones.

Flutes, clarinets, and misc. woodwinds

I like small omnis, but cardioids may also work fine too. Try miking from below the instrument to minimize breath noises, but this really requires a lot of trial and error experimenting to get right.

Concert and Celtic harps

Very difficult to get a good sound in a less than perfect room, but generally, a pair of X/Y small cardioids about 4 to 6 feet away, placed as high as the instrument is tall, and aimed slightly downward. Use your quietest mics, since these things don't put out a lot of sound and pedal noises can often be a problem. A couple of hints: Try putting the harpist (harpy?) in the corner of the room, facing out, to pick up a little more fullness without using eq. Also, try putting the harp on a 6 foot by 6 foot plywood sheet on the floor to help bounce a little more top end into the mics.

Misc. Percussion

For tambourines, cowbells, etc., a small omni is usually best, placed about 2 feet above the instrument. Watch your levels, and don't exceed -10dB, since most of the energy won't be shown on your meters. For misc. drums, try a large diaphragm dynamic mic, or a small cardioid condenser, and try miking close to the top, and even try miking from below the drum.

Digeredoos, harmonicas, and other odd shit

A dig is like a single organ pipe - with weird ass mouth stuff attached. Use a small condenser cardioid slightly off-axis from the end of the thing and adjust the height to pick up more mouth action. Harmonica players usually have their own mic, but if not, try your cheapest dynamic into a small guitar amp with a little bit of distortion. For clean harmonica, try an omni, about 1 foot above the player and 6" in front, aimed straight down. For other odd shit, I usually reach for an omni or small cardioid to "capture the moment".

OK, what have I left out?

Chris F :

I know for a fact that I won't be able to match the equipment you probably use, but if you were recording a jazz piano trio, what mics would you use and where would you set them? I'm especially interested in the mic choices for piano (for the future - I'm happy with my 603's for now), and the placement on the drums. I'm still working on the acoustic bass thing, best sound so far has come from a dynamic mic about 9" in front of the bridge, believe it or not...I'm going with that until I find something better.

Harvey Gerst :

Chris, I think you already have it pretty well nailed. Put together the techniques you're using for piano, bass and drums and go for it. Try to avoid eq, compression and effects when recording, and then see what it needs when you mix down to stereo.

You'll probably want to thin out the upper bass of the upright bass a little during mixdown (around 250 to 400 Hz), and maybe add a little 12kHz sheen to the piano, but everything should sit pretty solid in the mix. Don't make the piano too wide. My mic choices would be the same as yours - small cardioids in X/Y, near the lip of the piano. Record a couple of minutes of the whole band playing, then listen to it back and make any adjustments as needed. When it sounds right, it is right.

h kuhn :

What has worked very good for my Flügelhorn is micing it with an mc012 omni, about 30-50cm from the bell with the mic perpendicular to the airstream. Sounds very mellow but not nasal.

Harvey Gerst :

Flugelhorn, eh? You mean a trumpet on steroids? When I recorded Buddy Childers jazz flugelhorn many years ago, I think I used an old Sony condenser mic, about 18"to 24" away, and at an angle to the bell, but don't hold me to it - that was a long time (and many drugs) ago.

h kuhn :

It's a shame, but I don't know even one recording of buddy childers (and I am holding a master degree in Jazz trumpet...)

Many horn players emit a lot of ugly noises while playing (grunning, breath noises, saliva coming out in little bubbles at the corners of their mouth...especially the Jazz players.) Unless they play very loud, these can be rather disturbing, even in a mix. As these noises are high pitched, angeling the mic away from the bell (and the players mouth) can help. This is at least what I found out. And the omni pattern helps to get some room reflections, makes it more "live" (hate artificial reverb...), and, as the mic is actually aiming to the side, still captuiring the direct sound.

Harvey Gerst :

Buddy could play very quietly, with incredible tone.

h kuhn :

BUddy Childers, the lead player of the Kenton Band??? I am shure that he had a lot of control over his horn!

Harvey Gerst :

Yeah, that Buddy Childers. We were pretty good friends back then (I'd stay at his house in Vegas, and he stayed with me for a while when he moved back to L.A.).

Before I got into folk and rock, I was pretty heavy into the L.A. jazz scene, hanging out with Buddy, Freddie Gruber, Paul Horn, Ornette Coleman, Charlie Haden, Don Ellis, and all their friends. Hell, my first guitar teacher (at $5 an hour) was Barney Kessel.

JerryD :

Harvey, have you covered acoustic guitars in this thread? I'm not seeing it. Is there another thread that you covered this in?

Harvey Gerst :

I've covered some of the basic theory in this thread of why miking an acoustic guitar is so difficult, but I guess I should go into some techniques that are a little more specific. I'm about to post miking an electric guitar in a few minutes (I just finished the drawing for that), and then I'll see if I can sum up acoustic guitar miking next.

Ok, let's mic an electric guitar.

This is gonna be another very long post, so hang in there. I'll try and keep all the techie stuff to a minimum, but there are some concepts that are kinda hard to explain without getting a little technical, so ask questions if you don't understand something - it's probably just due to my poor explanation. Before we get into the miking part, we hafta talk about how speakers radiate sound, so here comes the first drawing:

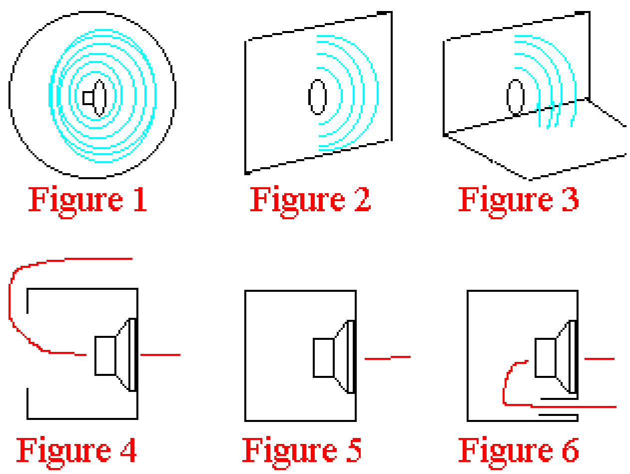


Figure 1 Figure 2 Figure 3

Figure 4 Figure 5 Figure 6

Figure 1. Imagine a speaker suspended in space - the sound comes off the front of the cone, AND off the back of the cone, more or less equally. The problem with this kind of setup is that the low notes coming off the back side of the speaker cancel the low notes coming off the front of the speaker. Their wavelength is much bigger than the diameter of the speaker and they just go around the frame easily.

Figure 2. Now imagine we've mounted the speaker in the exact center of a huge board 40 feet wide by 40 feet tall. The speaker is still radiating in all directions, but unless a low note is at least 20 feet long, it ain't gonna get around the edge of that board easily. Since we eliminated the possibility of cancellations, the bass comes way up when your standing in front of the speaker, compared to the speaker that was just hanging there on a string. As far as we've concerned, it's now radiating into a hemisphere.

Figure 3. Now put the speaker down low on the board and imagine a floor has been added. What happens? The bass notes double in colume since they're now radiating into one half of a hemisphere. If you put the speaker at the juction of the floor and two walls (a corner), the bass would double again, since all the bass is now radiating into a quarter of a hemisphere. But what does this hafta do with miking an electric guitar? You're about to find out right now.

Figure 4. If we fold the board (shown in Figure 2.) into an open-backed box, we can still prevent a lot of the bass from wrapping around and cancelling out. Starting to get it? Bingo, you're basically looking at a side view of most open backed guitar cabinets, like a Fender Twin. The box prevents some of the low notes coming off the back of the speaker cone from getting around to the front and interferring with the notes coming off the front of the speaker. This arrangement works ok till you get down to around 90 - 120 Hz, and below, right at the nottom end range of a guitar. So how do we get a little more bottom end?

Figure 5. Make the box a little bigger and seal it completely. Now the back notes can't interfere. Recognize the design? A Marshall cabinet? Right!!!

Figure 6. As long as we've come this far, I threw this in. You take the sealed box, cut a hole in it, and then you can tune the air in the cabinet to create a "blowing across a Coke bottle" effect, to add some bottom where the speaker starts to give out.

Keep some of this in mind when I start this next section:

Miking the guitar cabinet.

Guitar amps come in many different configurations, but I'm gonna focus on miking the three most popular speaker designs:

Open back cabinet, single speaker.

Open back cabinet, dual speakers

Closed back cabinet, with 4 speakers.

Here comes another one of those damn drawings:

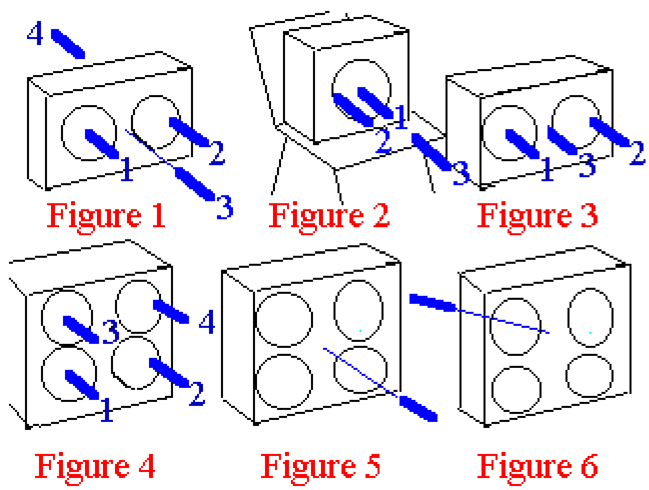


Figure 1. The two-12" open back speaker combo is one of the most popular units of all time. There are 4 basic mic positions, with several variations:

1. Stick a mic right into the speaker, aimed at the center of the cone. Maximum high end, and least outside noise.

2. Stick a mic right into the speaker, aimed at the edge of the cone. Less high end, and a little more bass.

3. Pull back a bit (12 to 24") and aim a mic right between the speakers. More realistic, but inreased chance of phasing problems and more susceptable to room noise.

4. Use any of the first 3 methods and add a mic aimed at the back of the speaker. Try the phase switch and choose the position that sounds best to you.

What mic to use?

Try the Shure SM-57, or your kickdrum mic, or any good dynamic for positions 1 and 2. Positions 3 and 4 might use a ribbon or condenser mic to get a little fatter sound. I usually start with position 1 (one mic, pointed into the center of the cone), but I may add something like an AKG D122 on the outside edge of the other speaker to emphesize the bottom end a little.

Figure 2. The single speaker open back speaker cabinet is another popular design. The are same 4 basic mic positions are used:

1. Stick a mic right into the speaker, aimed at the center of the cone. Maximum high end, and least outside noise.

2. Stick a mic right into the speaker, aimed at the edge of the cone. Less high end, and a little more bass.

3. Pull back a bit (12 to 24") and aim a mic at the speaker. More realistic, but inreased chance of phasing problems and more susceptable to room noise.

4. Use any of the first 3 methods and add a mic aimed at the back of the speaker. Try the phase switch and choose the position that sounds best to you.

5. Repeat all 4 mic techniques, but put the amp on a bar stool or chair. Why? Go back to the very first drawing and look at Figure 3. By raising the amp, it now feeds into a hemisphere instead of a half hemisphere, lowering the bottom end a little. Pull the amp away from a wall for less bass, in closer to the wall for more bass. See how the first drawing is starting to fit in?

What mic to use?

Try the Shure SM-57, or your kickdrum mic, or any good dynamic for positions 1 or 2. Positions 3 and 4 might use a ribbon or condenser mic to get a little fatter sound. I usually start with position 1 (one mic, pointed into the center of the cone), but I may slide it till it's at the outside edge of the speaker to emphesize the bottom end a little.

Figure 3. is simply there to use up some space. I just thought it looked better with 6 drawings instead of 5.

Figure 4. is a standard 4x12 Marshall cabinet. You would use mic positions 1 and 2 for adjusting the high end relative to the bottom end (and remember, you're getting that 1/2 hemisphere bass boost from the floor). To lower some of the bottom end, move the mics to positions 3 and 4 (or try a 57 at position 3 AND a D112 at position 2, then blend them to one track, or record them wide apart to two tracks).

Figure 5. Marshall cabinet with distant miking. Try a ribbon mic, or a big condenser mic to get a fuller sound. Adjust the mic anywhere from about 2 to 10 feet away. If needed, also use one of the mic techniques in Figure 4.

Figure 6. Actually this one is for any cabinet. Scenario: The guitar player isn't happy with any of the mic setups you've tried so far. Have the guitar player play with the controls till he's happy with the sound. Tell him to freeze, right there. Put a mic close to his ear, pointed at the center of the cabinet, and go back and listen. Either an omni, small cardioid (dynamic or condenser), or a large cardioid will usually do fine. The mic is now hearing "exactly" what the guitar player heard in the room when he said he liked the sound. That should end any conflict.

Hey, we're nearing the end of this whole mess - just a few more things to clean up, and then we're done!!!

JerryD :

1. Why wouldn't you put the dual speaker with the open back end into a chair. Wouldn't it have the same problems as the other cabinet configurations?

2. Why wouldn't you leave the single speaker config on the floor so it would have more bass. I guess my question is why are you trying to reduce the bass when the single speaker would have less bass and need to be increased?

3.

   Pull back a bit (12 to 24") and aim a mic right between the speakers. More realistic, but inreased chance of phasing problems and more susceptable to room noise.

   Could you give a brief explanation of phasing problems or point me to a article that clears this problem up for me?

Harvey Gerst :

1. Good call!! Yes, moving the speaker off the floor can help an overly heavy bottom end and get the speaker sound closer to what the guitarist is hearing, once he adjusts the tone to compensate for the movement. And it's an alternative position for any kind of speaker cabinet, not just a single speaker setup.

2. Most players don't realize how much boost they're getting from the floor. In many cases, there's simply too much bottom end and it interfers with everything from the kick all the way to the main vocals.

3. Pretty simple really. When miking a multiple speaker cabinet from a distance (rather than right at the cone), there is a chance that you might get some comb filtering caused by phase differences between the different path lengths from the mic to each of the multiple speakers.

On the good side, this is one of the most benign distortions there is. The old Bozak and Bose speakers used multiple drivers and nobody really ever complained about them.There are probably some old AES papers on multiple driver phase cancellations somewhere, but I don't know of any on the net.

Tekker :

Jerry D, I here are a couple definitions that I found; I couldn't find much in the way of really good articles/pages on phase (cancellation), most of what I found was just little pieces hear and there, kinda mentioning it but nothing really spetacular. Oh well, I hope these definitions help anyway.

Phase ("In-phase/Out-of-phase") Actually refers to the polarity of an electrical or acoustic signal. If two or more signals or devices are "out-of-phase" with respect to each other, cancellation or other disturbance of the combined output can result. Some examples of operating phase (polarity) definitions: "Positive sound pressure causes positive output" (microphones) or "positive input causes positive acoustic output" (speakers).

Phase Cancellation Undesirable dips and peaks in frequency response caused by mixing the outputs of two microphones which are picking up the same sound but with different arrival times. For example, this can occur when two microphones are placed near each other, but still with space between them; or when wireless microphone users stand next to each other.

blinddogblues :

I would have never thought of using my kick drum mic to record electric guitar, that blows me away. I just bought a Shure Beta 52. I'm going to try it, I will let you know how it goes.

I gave it a quick test and it sounds pretty good, micing my deluxe reverb with a Beta 52. I didn't put the amp up on a chair, but I did lean in back against the couch so it pointed upward at an angle. I stuck the mic right on the grill cloth pointed at the center of the speaker. Usually I use a 57 over at the edge of the speaker. It came out real good, especially on rhtyhm chords and on the bass strings (no surprise), but I played some solo notes in the high range and it sounds good on that too, even though it does not quite have the bite that I get from the 57. I guess it just shows there is more than one way to skin a cat. Perhaps this would be a good way to record a rhythm track, and then go back and use a 57 to record a lead guitar track?

Harvey Gerst :

Bingo, ya got it!! Except you might wanna record the rhythm track, move to an unused track and record a second rhythm track, pan them hard left and right, then switch to the 57 and record the solo straight down the middle. That gets the rhythm tracks really beefy, but out of the way of the solo and vocals.

h kuhn :

stereo

Harvey, do you think you might find the time to elaborate more the different stereo micing techniques and mic choices? And for what applications would you use them?

Harvey Gerst :

Sure, there are 3 basic stereo techniques, and several variations within each technique:

1. **Co-Incidence**
   These include X/Y, Blumlien Pair, M-S, and Phased Arrays
2. **Near Co-Incidence**
   These include Jecklin Disk, ORTF, Binaural, Crossed X/Y, and some others that I'm forgetting right now.
3. **Wide Spaced**.
   Omnis (or cardioids), spaced far apart, with center fills sometimes.

Whoops, I'm gonna need pictures for this. Hang in there, it's gonna take me a while. I'll ty to get them drawn up by tomorrow.

trumpetman :

recording trumpet

I record a lot of vocals and guitar (acoustic and electric), so I have learned a lot here I can use. However, my primary interest is recording trumpet, which I play professionally, so any more info on capturing that sound is welcomed. I play on various instruments, a lot of piccolo trumpet in particular, and I wonder if that instrument demands special equipment or tricks to be recorded well.

h kuhn :

I suppose you are playing classical music on the piccolo? try omnis and record in the best room you can afford. Try a pair of spaced omnis. If the room has a lot of reverb (church etc) get closer or use cardioids. Maybe give earthworks a try.

tubedude :

Harvey... did you skip drum kit recording, or did I manage to miss it somehow?
I actually have been reading around this post for quite awhile now, because when I read the 1st few, I've not been in a "technical" frame of mind. Started last night, and I must say, this is good stuff.

Keep it coming... more on the weird stereo techniques. That blumlein disk thing... explain that.. and all the other stuff.

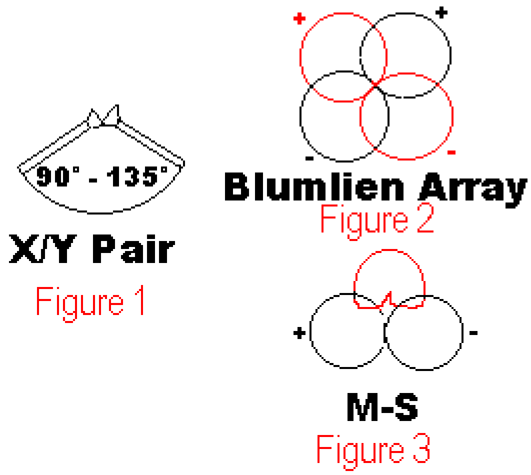
Do you know anything about phase-compensated stereo imaging and how it works?

Awesome work... what abour preamps and frequency carving?

Harvey Gerst :

I'm gonna break this into sections and then post the basics first, then talk about how and when to use them.

Coincident stereo mic techniques.



In the late 1920's and early 1930's, Alan Blumlein (in Britain) and RCA were both working on recording techniques, using only a small number of recording channels for reproduction over a pair of loudspeakers.

The technique developed by Alan Blumlein consisted of a pair of microphones with figure 8 patterns, mounted close together,with the front lobe of one mic pointing 45 degrees to the left, and the front lobe of the other pointing 45 degrees to the right (Figure 2.). Although it provided excellent stereo imagining, sounds coming from the rear are also picked up and when reproduced over a pair of loudspeakers, these sounds were also mixed into the speakers. This results in a sound which is too reveberant for many people.

"Purists" who liked the simplicity and accuracy of the Blumlein technique modified it in order to remove this problem. By replacing the figure 8 microphones with cardioids and changing the angle between them to include the desired soundstage, it is possible to use the cardioid mic's lack of rear response to reduce the rear reveberant sounds. This results in a much more acceptable, if less accurate sound image. Typically, the angle between the mics should not be more than about 135 degrees, or less than 90. This technique is the popular "X/Y" stereo recording system. (Figure 1.)

Mid-Side (M-S) techniques use a figure 8 mic, sideways to the sound source, and a cardioid mic facing the source (Figure 3.). By inverting the signal from the real lobe of the figure 8 mic and using a matrix network, it is possible to adjust the width of the sound stage to almost any size.

JerryD :

1. In the X/Y pair I'm assuming that the mics have to be identical. Correct?
2. In the X/Y pair where and how far away would the sound source be?

3. You lost me on the Matrix network. What is that?

Harvey Gerst :

1. Yes, it helps keep the stereo image from wandering. (That means if one mic has a peak at a certain frequency, everytime that note comes along, the mic will hear it louder and play back as if the sound is coming from one side.)

2. Depends on the X/Y angle you use, but in general, no further back than the mics pointed at the outside edges of whatever you're recording. At 90°, the mics would form the apex of an triangle, with one mic pointed toward the left edge of the group you're recording and the other mic etc., etc.. As you move the mics in closer to the source, you would widen the angle accordingly. it's not a hard and fast rule. For example, I usually aim the mics about 1/4 of the way in from the outside of the source, and then move in and out till I get the sound I want.

3. Ok, this M-S stuff works by additive/subtractive matrixing. True figure 8 mics have the best off axis response of any pressure gradient design and a perfect null in the middle (at exactly right angles to the front of the mic. When you turn a figure 8 mic sideways to the sound, none of the sound from dead center is heard. Only from the left side, or the right side (but one side is out of phase, cuz it's coming off the back of the mic).

You add a cardioid pointed at the center of the music. and not only does it fill in the hole, but it is combined with the figure 8 to create sum and difference combinations (usually thru a matrixing box) to let you control the absolute width of the stereo image. You can dial in anything from a perfect mono signal to wide stereo, all with perfect phase coherency.

A more complete source for how it does this is available from my friend Wes Dooley at http://www.wesdooley.com./ He makes M-S matrix boxes and he has a complete article on how M-S stereo works on his web site. Wes is one of the leading authorities on M-S Stereo miking. (I can't afford any of his damn matrix boxes, but it's good reading to understand the principles.

JerryD :

4. On the X/Y pair it looks like if you go over 90 degrees on your angle you would start missing the center of your stereo image due to the limitation of the mic pickup pattern. Why would you ever want to go over 90 degrees?

Harvey Gerst :

4. To increase the apparent width of the sound stage. But, as you widen the angle between the mics, you move in closer to the source, and you move in closer to the center, more than you do to the ends of the source, so the center level inreases to offset the loss from the wider angle. In simpler terms, it all works out.

gordone :

M-S Question

I've been recording some acoustic guitar (fingerstyle) and I tried what I think might be sort of M/S micing. I have a Soundelux U97 (Figure-8) and a KM184 into a Great River MP2. I point the U97 sideways a bit to the left of the soundhole (~8" out), and the KM184 pointed towards to 10th fret about the same distance out...I've been pretty happy with the sound but I want to make sure I'm not missing something. Am I supposed to have the mics on top of eachother and pointing at about the same spot on the guitar?

Also, I don't have any matrixing stuff, but I record to my PC and use N-Track, I believe there are plug-ins that can reverse the phase of a channel, but could you refresh my memory as to how to do this, and also let me know what I'm missing by not doing any matrixing of the two channels.

electricbeats :

M - S

Maybe I can help you out on that too but prior to it please some mor infos:

What's your main goal using M - S ?

Since M - S is a sterephonic technique its advantage is not just the natural image of a sound source in a sereophonic field but also the - if done right - nearly absolute mono compatibility it is quite unusual to close mic an instrument with it.

The second thing would be:

You said you had one mic placed at the 10th fret the other one at the soundhole - this is not M - S. Ideally the mics have to be exactly at the same position (in theory). What kind of stereophonic image you get depends on the polar patterns you use. cardioid, hyper, 8 ...

Specially in close miking in that technique is essential to have both mics at 'the same place' since otherwise you R suffering combfilter efffects...

The way you place the mics seems to me that you want to derive the stereo effect by miking the instrument on two different positions with the result of different freq response in the two channels which is perfectly right for what it is, just that it's not M - S

Harvey Gerst :

We still have a ways ta go yet. I've just been too busy these last couple of weeks and it looks like it'll be another week before I can concentrate on this stuff again.

I'm in the middle of moving into a new house, getting a master out on a project I've put a year of work into, expanding the studio by adding a second larger room, buying a new (well, used but still expensive) new board, trying to finish up the Studio Projects mic tests, all the while recording bands on our normal schedule. It's been a little crazy here, and it'll probably stay that way at least untill next week or so.

And electricbeats is right; what Gordone is doing may sound nice, but it's not M-S stereo recording.

GrooveBassman :

Ok Nearfield amd farfield. So why does say an instrument sound in front of the mix if it's miked up close as opposed to distant. It 's just that the closer the instrument is to the sound source the more in front of the mix it sounds? Vocals close up stand out guitars and toms sound in your face. Is this just a mental thing or is there an acoustic reason close miking cuts through a mix.

Kelly Holdridge :

I think that's because there's less room sound, and the brain can subconsciously pick up (from presence/lack of room) how close each instrument is. Even in a mix.

electricbeats :

near/farfield

The room/source relationship to position an instrument in the mix is essential to the position in a mix. Therefore instruments/vocals that have to hit (right between the listeners eyes) are recorded in a almost dead booth (no earyreflections/reverb) , acoustical instruments like drums, strings etc are recorded usually in a 'live' room (earlyreflections depending on size/ reverb depending on surface and the relationship width/depth/higth) the room component defines just the depth but not in the stereo field.

To verify this just listen to a classical recording - the component of the (acoustically designed) room is high there - then listen to a typical nineties production. You will find out that a vocal track sitll hits you right in your face sometimes includes quite long reverb tail. Normal reverb would bring the voice far to much back in the mix. The answer to that is pitch shift to double up your vocals. I know that this is not topic here, all I'm trying to say is that you should not try to compare nowadays vocal tracks' position in the mix just by the room component (artificial or natural) - listen a lot to classical music (operas!) --- OR try to record your voice in your living room (once close/once far) - in your bathroom and in a church (w. MDwalkman?)- and then in a booth or any small room that 'kills' the trflections- maybe then record with your lex or yamaha.

Thats the only way to find out whats going on. its interesting and worth spending some time on it...

Harvey Gerst :

OK, let's finish up stereo mic techniques and move on.

I'm not gonna go thru all the pictures with circles and arrows this ,since most of these techniques are listed in detail with pictures at the DPA Microphone University site at:

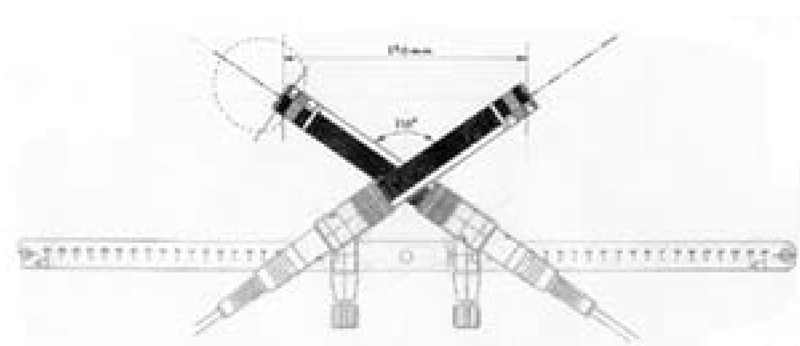
http://www.dpamicrophones.com/46a.htm

But we'll discuss all of the techniques here. We've already covered "coincident" stereo mic techniques (M-S, X/Y, etc.), all of which involve stereo from phase differences between the sound sources. The mic diaphragms are spaces close together, and the angle between the mics is used to control the phase differences.

Near Coincidence Stereo Techniques:

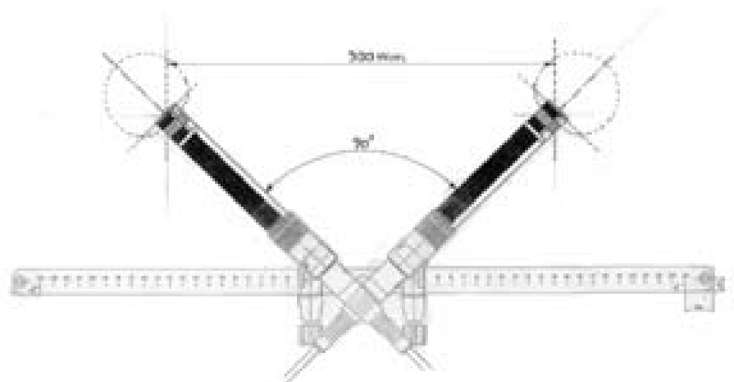
These use time and level differences between the two mics to achieve the stereo effect. The mic diaphragms are spaced at different distances or something is put in between them to approximate the way the human ear hears stereo. The near coincidence methods include:

ORTF (France): A pair of cardioids at 90° (pointed away from each other), spaced 17 cm apart. Looks like this: \ / (except the actual angle is 90°.

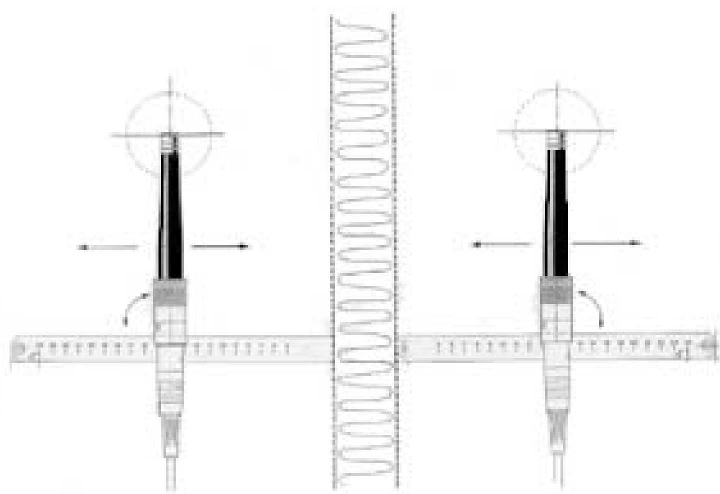


DIN (Dutch): A pair of cardioids at 90° (pointed away from each other), spaced 20 cm apart. Looks like this: \ / (except the actual angle is 90°.

NOS (Netherlands): A pair of cardioids at 90° (pointed away from each other), spaced 30 cm apart. Looks like this: \ / (except the actual angle is 90°.



Jecklin: A pair of omnis spaced 36 cm apart, with a disk between them, like this:

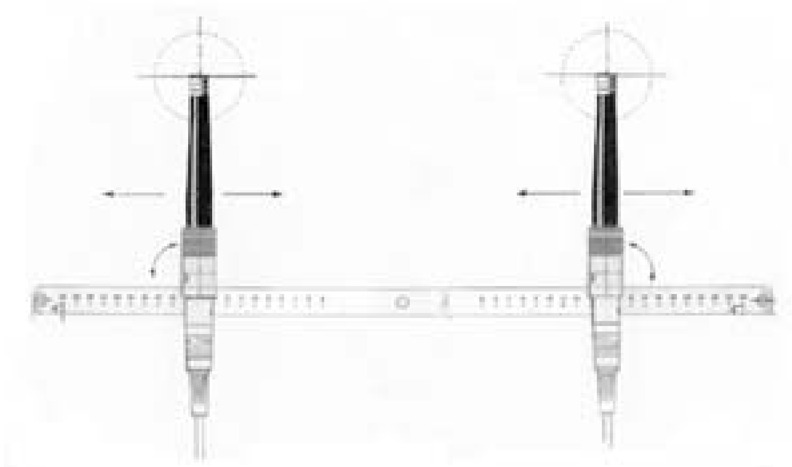


Binaural: A pair of omnis, usually mounted in an device which approximates the dimensions a human head.

All of these near coincidence techniques depend on time and level differences to get the stereo image. Which you would use depends on the source of the sound you want to capture, the room and equipment you are using, and the accuracy of the stereo image desired.

A-B (Wide-Spaced Stereo Techniques):

A-B techniques all use time and/or level differences to get the stereo effect. For large groups, a third (center) mic is often used. The standard A-B miking uses a pair of omnis spaced about 40 to 60 cm apart, facing the sound source. Cardioid can often be used instead, and a third mic in the center can be added to fill in any hole the wide spacing causes.



Stereo Wrap-Up Thoughts:

The effect of all these different stereo miking techniques will vary, depending on whether playback is for speakers or for phones, and the width of the source being recorded.

Generally, you'll get more accurate stereo effect using the coincidence or near coincidence methods, especially for smaller single instruments or small groups, like a string quartet or bluegrass band.

A-B wide spacing usually works better for loudspeakers and wide sound sources, like choirs, orhestras, and pipe organs. But M-S, Blumlien, Jecklin, and binaural can also work well on large sources.

All these stereo techniques are time-tested tools that work well on a lot of different things. As the person doing the recording, it's important that you understand how they work.

The choice of which to use for a given project is up to you, but now, you at least know all of the techniques that are normally used on a lot of records that you've heard.

Still to come: Drum miking, piano miking, percussion miking, and how to read (and understand) mic specifications.

tubedude :

Ribbon Mics

Harvey, a big part of my interest last few months or so has been in ribbon mics. How are they better/worse, etc. I hear they make great room mics, overheads, and get a huge dirty guitar sound. Maybe its just the Royers. Been wanting to check out the Beyer ribbons and find some comparisons against the Royers.

Harvey Gerst :

Paul, ribbon mics are a wonderfully simple, yet elegant, solution to a number of mic design problems. Using a single strand of superlight corrugated aluminum ribbon, they eliminate a large part of the resonant, transient, and axial difficulties of other microphone designs. They have the disadvantage of being fragile, having very low output, susceptable to hum fields and wind blasts, and the older ribbon mic designs were heavy as hell.

I own four ribbon mics, and I'd sell almost everything else in the studio BEFORE I'd part with them.

A few more thoughts about stereo miking.

First, I want to add a few pointers about stereo miking that might help clarify some terms for people that are new to recording.

Coincident Mic Techniques

That's a fancy term for any stereo mic technique where the two matched capsules are as close together as possible and only the phase differences are used to get the stereo image. The advantage to these techniques is that they collapse to mono very well, sound good on speakers, and amazing on headphones.

Near-Coincident Mic Techniques

Another fancy term for any stereo mic technique where two matched cardioid (or omni for some techniques) capsules are wider apart, pointed parallel (or away from each other) and use level differences and time delays to get the stereo image. The different spacings determine the stereo image spead. The advantage to these techniques is that they sound excellent on speakers, and very good on headphones.

Wide Spacing Mic Techniques

A term for any stereo mic technique where the two or more similar cardioid or omni capsules are wide apart, aimed parallel to each other(or angled in) and use level differences and time delays to get the stereo image. The advantage to this technique is that the sound is amazing on speakers, but not very good on headphones.

So which technique should you use for your recording? It depends on what you're recording, and what part it plays in the final recording. If it's an acoustic guitar part for a full rock band, forget about stereo and just use one mic, usually. For a singer/songwriter, try M-S or coincidence recording.

If the recording might get a lot of FM radio play, coincidence recording will give the best mono signal when you get into weak signal areas. String quartets, bluegrass groups, barbersho quartets might work better with near-coincidence mic techniques, while all three techniques might work well for larger groups.

You hafta ask yourself some question when choosing a stereo mic technique; How important is mono?; how much spread do I want; how important is one element in the stereo field (like a soloist in a choir)?

Like everything else in recording, experience comes with time, experimentation, and failures. We learn from our mistakes. And sometimes, our mistakes save our asses. Record, listen, then analyse:

What's missing? Why?

What's too loud? Why?

What's too boomy? Why?

What's too muffled? Why?

If you constantly hafta reach for eq to correct things, it usually means that the mic is:

1. in the wrong place, or
2. the wrong mic for the job.

Any more questions about stereo recording techniques, or does this about cover the subject.

h kuhn :

Yes! Why is it that in classical recordings the mics are very often set up above the orchestra/ensamble, while the audience normaly sits below? wouldn't it be more logical to put them where the listeners ears is?
The other question: how important is mono compatibility nowadays? I mean, how many % of the population (even in 3rd world countries) uses a mono playback device? I haven't seen one since many years.

Harvey Gerst :

a) Many orchestras DO put two mics in the listener's position, but it often captures too much of the hall's reverberation, so they use spot mics to cover different sections of the orchestra, and then they blend the various sections together during mixdown from a multi-track recorder.

b) Mono capability is still important in two separate areas - Television, and FM radio. There are still a lot of TVs out there with only one mono capability. And when you get into a lower signal strength area, your FM radio automatically switches to "mono receive" mode.

c) Now, let's say you recorded your rhythm guitar track on the left channel, and then ran the same track thru a short delay to the right channel - to make the guitar sound bigger (a common home studio technique). It'll sound fine in stereo, but the guitar will DISSAPPEAR COMPLETELY when summed to mono.

h kuhn :

a) But have you ever seen a decca tree at listeners position? As far as i know they are usually hung about 2m above the conducters head, aren't they?

b) OK, but would you avoid to use a spaced pair of omnis to record an orquestra if you knew that the concert will be transmitted on TV? Or maybe I should ask: Can you achieve satisfactory mono-kompatibility (what a word) with that kind of setup? Or would you prefer a coincident technique, even if the soundstage doesn't translate that well?

Harvey Gerst :

a) Yes, and that's mainly because the listener's position would have too much of the reverberant field present. You must also account for the fact that the end listener will probably be listening to the final product in their own reverberant field (i.e., their living room). The conductor's position would (or should) be the most balanced spot in the hall, since he controls the level of each section of the orchestra.

b) When you hafta get it right quickly, you go the safest possible route, which means a coincident pair, if you're restricted to using two mics and you need a recording in a hurry without a soundcheck beforehand. M-S would allow you to adjust the soundstage in post production, and it would be mono compatible.

The "Making Of 'The Producers' Soundtrack" is going to be on PBS tonight and I'm looking forward to finding out how they recorded it. But, to answer your question, yes, a pair of spaced omnis would probably work fine, but most commercial orchestral recordings are usually made with each section miked.

h kuhn :

Harvey, I think I found a different answer (which coincides very well with what you said about radiation patterns): I recorded a violin player last week. The venue was a fairly big theatre, I decided to try xy from about 2m high and 3m away from the player. The acompanying piano (a 6ft steinway, sigh..) had the lid half closed and was directly behind the violinist. The violin sounded great from above and only so and so from aprox 1m height and the same distance, while the steinway sounded better. I guess that the violin is simply radiating upwards/sidewards, very similar to the cymbals of a drum set. And much of the orchestral sound comes from the strings, while they seem to be the weakest part concerning spl, so it makes sense to accentuate them a bit. Are these very stupid ideas or am i on the right track?

Harvey Gerst :

Harald, you're exactly on the right track. Now that you have a good idea about radiation patterns, some different positions open up to you and you begin thinking "outside the box" - a fancy term for avoiding most people's preconceived notions of how something should be done, based on limited experience, or just reading "how-to" books.

Recording Drums

Drums - the "king" of rock instruments; so many different sounds, so many different textures, so many different sources.

Where do we start? How many mics do you "really" need to record a set of drums? One? Two? Three? Four? Eight? More? And what are the best mics for drums? Ribbons? Condensors? Dynamics? Small mics? Big mics? Where do you put them for best sound?

How big a room is right? What's too small? Is there a such a thing as too big a room? If drums require a big room, why do many major studios use a small drum booth?

If some of these questions sound familiar to you (because you've asked them before), then this next section is for you. I'm gonna dispel some myths, and show you how to get a good drum sound in any room. It's gonna be a big section, but it has to be. It's similar to miking a guitar; classical guitar is different than steel string guitar is different than electric guitar.

We'll cover drum tuning, music styles, mic selection, room considerations, and mic placement - all in detail.

Drums, Here We Go.

This is gonna be a pretty big addition, so I'll break it up into several sections. Obviously, recording drums depends on a lot of different elements; the actual drums used, the drummer, the room, the style of music, the mics available, mic placement, number of tracks available, stereo or mono, and how important the drums are to the particular song. Let's look at each of the above elements in a little more detail (although I'm gonna go into a "lot of detail" about kick and snare right now):

The Drums

Bad drums will never sound great. The drums hafta be in good shape, tuned correctly, and properly set up.If they sound bad in the room, they'll probably sound bad on tape. A good engineer sometimes has to be a good intrument tech. I've had to tune drums many times, intonate guitars and basses, rewire pickups, etc. Just because the drummer knows how to play drums is no guarantee that he/she can tune them.

Every drum has a natural resonance. You can hear the note by lightly tapping on the side of the drum. That's usually what you tune the top head to, with the bottom head tuned a little lower. There's a range of about 2 or so notes each way from that natural resonant frequency that will work fine, but you need to stay in that range to get the power out of the drum. Drums are usually tuned in fourths, starting with the high tom. If you're not knowledgable about drum tuning, it would be well worth it to have a good drummer come in one time and show you how to tune drums.

I'll get into drum tuning in another post if anybody's interested in that, but right now, just make sure the drums are tuned correctly, and they sound good in the room. We usually use Ambassador coated heads for our drums and they record very well. We avoid the oil-filled heads (too dead-sounding), and we stick with the single ply, coated top heads for everything, with clear heads on the bottom.

The Kick Drum

The kick, along with the snare and electric bass, is usually the backbone of the song - these instruments provide the "groove" and "drive" of most rock music, and they require the greatest attention. For rock, especially metal, the kick also provides another element - the beater "click", needed to hear the speed or complexity of the bass drum patterns.

Most rock drummers have a hole cut in the front head (the head facing the audience), but few drummers understand the hole's function. Most do it for looks, because "all the other drummers do it".

The hole is for mic access to the back drum head (the head being hit by the foot pedal), to let the mic get close enough to pick up more of the beater "click". The hole should be 4 to 6" in diameter, and located above the center line, to make it easy to get the mic (mounted on a short stand and boom arm) inside the drum.

I usually have the drummer loosen and turn the head till the hole is in the upper right quadrant, and I'll bring the mic in, angled toward the floor tom, about 3 to 4" away from where the beater hits the head. Angling the mic towards the floor tom reduces the amount of snare bleed, which will help later on if I need to gate the kick drum.

For drums without the access port, I'll also try miking the kick from the pedal side of the drum. If I need a really "huge" sounding kick, I'll construct a tunnel from a packing blanket off the ported head and add a large condenser or ribbon mic about 3 to 4 feet away (inside the "tunnel"), just to pick up the low end. I've even made a "tunnel" by removing the front head entirely and placing a second kick drum in front of the first (removing the back head from the second kick, and miking the second kick at the hole.

I avoid gating or compressing the kick during the recording stage, but I might do it during the mix. I usually add a few dB of boost between 2 and 4 kHz to emphasize the beater click. I'll crank the boost all the way up, and then sweep till I find the desired click sound, then back off on the boost. For tape based systems, this should be standard procedure, since boosting top end later on will also add hiss.

I'll scoop out a big hole down low, using a parametric, anywhere from 250 to about 800 Hz, eleminating the "boom" frequencies. I don't usually add any low bottom boost during recording, since it's easy to add later during mixdown.

The Mics For Kick Drums

For rock drums, the mic choices are usually: AKG D-112 (older recordings use the AKG D-12E), ElectroVoice RE-20, AudioTechnica ATM-25, Sennheiser 421, Shure Beta 52, and the new Sennheiser 602. These are all dynamic mics, either cardioid or hypercardioid, and pretty large diaphragms. The Shure SM-57 is also used for kick, and will work ok, but not usually as well as the mics listed above.

Mics for use in a kick tunnel or for distant kick miking are usually either large diaphragm condensers (the Neumann U-47fet is the most popular choice), or ribbon mics like the Royer 121, the RCA 44BX, or the Coles 4038 - all high dollar mics. Some good low-cost choices would be the Marshall V67G and the Studio Projects C1.

That should do it for a while. I'll continue the rest of this a little later.

tubedude :

I build my kick tunnels out of couch cushions and throw pillows. For some reason, it gave me an odd idea that I think I might try as a weird effect, with lots of boomy spacy reverb, maybe sample it for use later... turn a 55 gallon drum sideways with the lid off, and put a kick beater to it, with a mic at the end.

Maybe a double kick pedal and have the drummer do some wild triple beats and shit on it. If you think thats weird, well... go see the blue man group. Bad ass. http://www.bluemangroup.com/

Harvey Gerst :

Recording drums - Things to think about

I have some time so here goes the start of this section:

Drums: One **instrument**?

There are two schools of thought on this. Since each drum and cymbal basically produces just one note each, it may be thought of as simply one large instrument.

You can mic a drum set with just one mic, but it's tricky. You pretty much move the mic around till you find the right balance between the snare, toms, kick, cymbals, and high hats. That's usually about 6 to 8 feet away, and about 6 to 8 feet up in the air.

BUT that means you're also picking up a lot of the room, and if you have a shitty room, the drums won't sound all that great. So how do you get around that? It's actually the same problem as miking an acoustic guitar or a grand piano. Move into the instrument's near field (get closer), but when you do that, all bets are off.

In the near field, you have to use more mics, watch for phase problems, and realize you're gonna hear resonances a lot louder than normal. To keep phasing problems down, try just two mics (above the drummer's head, aimed at each end of the drum kit), then see what needs more oomph. You may need to add a snare mic, or a mic on the kick drum, but at least you won't be fighting the room.

If you have a lot of available tracks, you don't even need to commit to a particular drum balance right now - just put a close mic on every tom, the snare, kick, even the high hats, and worry about balancing everything out at the mixdown. (We'll talk about phasing problems that can occur in a little while.)

The snare and kick - the heart of the drum set. Ok, so it has two hearts. In actual fact, the snare is the heart of the set with the kick a close second. Everything revolves around the snare. When you set your overhead mics to pick up the cymbals, use a tape measure so that each mic is **exactly** the same distance from the center of the snare head.

Miking the snare up close - decisions, decisions. I use a hypercardioid mic (the Beyer M201) to mic snares, since it has some nulls at 130° off axis, so I can angle it to reduce hihat bleed into the snare mic channel, but even the Shure SM57 kills for snare if placed right.

One of the first places I try is the spot between the high tom and the hihat, aimed at the center of the snare, about 1" above and 1" inside the snare rim. If that doesn't sound good, I then check the actual sound of the snare to make sure it's tuned right and not creating a lot of problems. Sometimes, a little butterfly of duct tape on the snare head, right in front of the mic, will reduce ringing and spurious resonances enough to get a usable sound.

If that doesn't work, I'll check different mic placements, even to the point of pulling the mic back a few inches and moving it up and down the height of the shell, looking for a good balance (yes, actually pointing the mic at the shell, not the snare head).

Finding the right place for the snare mic can actually take upwards of an hour, but it's well worth the time spent. Once I have the snare sound about 80% nailed, I'll go to the eq and do any trimming that's needed, roll off some bottom, add a little mid crack, or some high end.

I usually add some short plate reverb to the sound of the snare, even if it's just in the headphones for now. I don't get "super anal" about the final sound, since I know it'll hafta change a little bit when I have all the other instruments mixed in.

Then I move on to the kick, which I'll cover in the next post on drums.

Some of you may not be familiar with the duct tape "Butterfly" used on drums. I've (hopefully) attached a picture that explains it a little better. It use a piece of duct tape shaped a little like an upside down butterfy to use for damping out any nasty head resonances.

It works better than a flat piece of duct tape, since the footprint can be smaller and it adds more mass in a given space, so that it damps just the problem spot without affecting the surrounding area. It can be as little as 1" wide and 3/4" high.

Any questions?

mixsit :

Harvey, when you refered to the two overhead mics being 'aimed at each end of the kit', did you mean front and rear?

In other references to a 'three mic tecnique', the primary mics seem to be directed as much front/back as left/right. If that's the case, would this call for more of a mono/center blend mix on the drums, rather than paned? It could be I missunderstood altogether!

Harvey Gerst :

Actually, I meant left to right. Think of a plastic dome over the snare, with the center of the dome being the center of the snare head. (Or think of it as a force field around the drum kit, with the center of the snare head being Voyager, or the Enterprise).

Think of the overhead mics as "photon torpedos". You want to place the two overhead mics so that they are at the same distance from the snare (like touching the surface of the dome). Now without moving the front of the two overhead mics. you angle the mics outward, so that they are aimed more toward the outside edges of the kit.

I usually aim one of the overheads at a spot between the high tom, snare, crash, and high hat, and the other mic is aimed at a spot that sees the floor tom and the front edge of the ride cymbal.

But the business end of each overhead mic is equidistant from the snare.

A few more thoughts to wrap up drum miking

Before we move on the last section (understanding mic specs), I want to finish up this section with some tips, tricks, and general thoughts about drum miking:

You don't always need "stereo drums". Unless there are a lot of tom rolls (or the drums play a huge part in the song), sometimes mono drums will work better to preserve the mood of the song.

If you DO mix the drums in stereo, you don't always need to pan everything wide. Closing up the stereo width will help bring the set together. Most drum sets aren't as wide as the distance between your home stereo speakers.

GENERALLY SPEAKING, you use large diaphragm dynamic mics on the drums, and you use small diaphragm condenser mics to bring out the detail of the cymbals.

A large condenser room mic can sometimes help bring the whole drum set into focus - if you have a decent room. Works best at 6 to 20 feet away.

The better the room, the less mics you need.

You "close mic" drums for two reasons - either the room sucks, or you want more options later, during mixdown.

Avoid gating and compression during tracking. Use it during mixdown if you need to.

Avoid recording the drums with reverb or effects. Add those later if possible. If not, add a little plate reverb to the snare during recording and just a touch to the rest of the drums. Record the kick dry.

If you have enough tracks AND you're close miking, record each drum to a separate track. If you're track limited, record the kick to it's own track, the snare to it's own track, and the rest of the set to a stereo track.

Use eq to get into the ballpark of what you envision as the final drum sound, but don't add big amounts of bass boost at this point - you can add that later, during mixdown. The snare will be the hardest to get right, but getting all the toms to sound even will also be a big challenge.

You can get a bigger deeper kick sound by building a tunnel and adding a second mic about 4' to 6' out from the kick, in the tunnel.

You can also put a second kick drum in front of the first kick drum, and mic that - with or without heads.

You can run the whole drum mix thru a pair of speakers and mic the speakers and mix that in with the original mix to fatten the drums.

You can lay a speaker on top of the snare, mic the underside, and feed the snare track to that speaker to add body, more snare rattle, or change the sound of the snare.

You can run the drum mix thru two compressors to really fatten the drum mix - here's how:

Set the first compressor for maximum compression (20:1 or greater), set the threshold for about -8 dB, and set the attack and release pretty slow (a little past halfway - you'll need to experiment to find the right settings). That knocks down the really loud parts without touching the faster initial peaks.

Take the output of the first compressor and feed it into a second compressor. Set the second compressor for maximum compression (20:1 or greater), set the threshold for about -3 dB, and set the attack and release to their fastest settings. That trims the fast stuff and what you get back is a huge sounding drum kit. The output of the second compressor is your final drum sound.

Hopefully, this covers all the drum stuff, so the next section (the last section ?) will cover understanding mic specs, seperating the truth from the hype and BS, and how to really read a mic frequency response curve (how they're created, and what they REALLY mean).

Did I miss anything?

Henrik :

Ditto that! Harvey, it's really very generous of you to take your time writing all this. I can't wait to try out the double compressor-trick on drums!

However, I have two questions:

a) When you emphasize the importance of placing each overhead mic at exactly the same distance from the snare drum, is this to avoid phase anomalies, or is it because you think it's important to have the snare in the center of the stereo image?
b) If you mic a drum (or anything) with two mics at different distances, as you suggested for the kick drum in your previous post - what steps do you take to avoid phase cancellation (if any)?

Harvey Gerst :

a) It's primarily to avoid phase anomalies.

b) I try to make sure that each mic follows the 3:1 rule, mentioned earlier in this thread. (I've been bitten on the ass many times for this in the past.)

Henrik :

Ah, how could I forget the 3:1 rule. I've gathered all of your posts here in one single Word file so I can easily go back and read it again whenever I need it. So yes, this is no doubt very useful!

ausrock :

Harvey, are you intending to start a thread on drum tuning in the future, as threatened? I think it would be valuable in it's own right as even a lot of drummers don't really know how to tune their kits correctly.

Harvey Gerst :

You're right about some drummers not understanding how to tune their drums. I was going to get into a whole section of how to tune drums, but then, I found this web site that covers it pretty well:

http://www.drumweb.com/profsound.shtml

Kelly Holdridge :

raises hand

Ok, I've been going off of John Sayers' approach to drum micing, which includes putting the snare in the middle of the stereo overheads by offsetting the axis of the stereo spread. Here's a picture of what I understand John to be talking about : (*next page)*

Ok. What I'm wondering is why worry about phase cancellation with precise measurements if you can get the diaphragms close enough together in an X-Y config? Is it because of the middle tom (or whatever sits below this offcentered axis) being louder than anything else? Is there a distinct problem with this setup? (the snare and hi-hat both have the same off-axis position to the 57's)

Oh, and that's an RE-235 hanging from the ceiling, our only omni. Since omni's are... omnidirectional, is it okay to just hang them like that? I mean, without any axis to worry about, (and outside of tugging, etc.), this is acceptable, yes? (prolly should be further away; we use it as our harmonica mic, too, and that's where it stays inbetween uses)

And if anybody wants to answer the bonus question, here it is (live sound question, but for recording purposes): That vocal mic picks up more snare than the overheads do, a result of Brian (my drumming brother) harmonizing using falsetto, which isn't too loud. Any thoughts on a way to get more of him and less of the drums on his vocal track? (it's a "Sound Addict" mic. i dunno, don't ask) I've considered using a "foamed dog collar" approach to his mic, but that just reeks of disaster. How do drummers who sing while playing get mic'ed in the studio (he HAS to drum, btw).

Vox :

A couple things that might help, use a boom stand instead of a gooseneck so that you can point the mic directly at his mouth and point the back (or null point) of the mic at the snare, use a mic with a tight cardiod pattern.

Harvey Gerst :

Sorry, Kelly, I missed the bonus question first time around. Usually, we'll use a headset mic to record a scratch vocal, and then replace it with a final vocal after everything else is done.

gnarled :

Well, I think with the john sayers approach he's talking more about spaced pairs, not coincident pairs. With spaced pairs (refering to your picture) you might put one overhead over the drummer's left shoulder and the other one in front of the kit on the side with the crash and high rack tom. That way, both mics will get the snare at approximately the same level so it will be centered when panned and (if you measure the distance to the snare) the snare will be in phase.

Henrik :

Harvey, you mentioned some time ago that you would explain why many studios use small drum booths instead of nice sounding rooms. So why is that? They trust their digital reverbs and want the drums as dry as possible when tracking?

Harvey Gerst :

That's part of it, but mostly, it's to keep the drums from bleeding into all the other open mics.

Mixsit :

Thanks Harvey. Sorry for the delay, been out of town... So, if the mics are close together above the drummers head, you've got sort of a narrow XY stereo set up, except the mics piont out instead of across?

I definitley like the idea getting a little behind the kit. I used to do XY above the kit, but switched to each side near the drummers sholders when I went to Earth omis. It got me closer to the drums (skins), more isolation from the other instruments and farther from the cymbals. Almost never see the need to close mic the toms. The hat being a little too loud and left seems to be the main disadvantage (but that's usually because he's playing it too loud). In keeping with the mics being equa-distance from the snare, the left mic gets pulled back a bit which also helps.

Harvey Gerst :

I prefer using wide spaced omnis (Audix TR-40, MXL603S, or Oktava MC012s w/Omni capsules), or cardioids (Shure SM-81 or Oktava MC012s w/Cardioid capsules) for overheads. It gives me a feeling of greater control over the stereo image, but it's all about whatever works for the music.

One last note about drums before we move onto the last section of this whole mess:

I usually stand in the drum room and listen to the drummer and watch him for a while BEFORE I start picking or placing mics. I watch where he hits each piece of the set and try to establish a feeling of where the mics are gonna be out of the way, AND it helps me decide on which mics to use.

If he's heavy on the high hats, I'll position the overhead to pick up more crash, or switch him to a 12" set of high hats to soften them. Heavy on the snare? Pull the snare mic back a little (and maybe angle the kick mic away from the snare side a little more). Light on the snare? I'll move in closer. Lots of tom rolls? I'll double check the sound of the tom mics against the overheads for possible phasing problems. Lot's of light cymbal work? I might move the overheads in a little closer to pick up the delicacy and shimmer.

But really listening and understanding how the drummer approaches the drum set is very important to getting the right sound.

Ready for the next part?

OK, if we're done with drums, we can move on to understanding mic specs (and I promise you that there will be a lot of surprises there). Polar patterns are NOT what they "appear to be", "fudge factors" are major parts of most spec sheets, and microphone frequency response curves are often meaningless, misleading, or worthless, either by accident or by intent.

Harvey Gerst :

Microphone specs ("I don't think we're in Kansas anymore, Toto")

I know a lot of you read the specs of mics very carefully to help you make decisions before buying, but that could cause you some serious problems. Here's why:

Let's just look at two manufacturers that are highly acclaimed here; Marshall and Studio Projects. Keep in mind that I think very highly of both these lines, but their web pages show some serious inconsistancies. For example:

If you go to http://www.mxlmics.com/ and you want to find out about the 1000 series, you click on it and up comes a nice, very impressive specs page, complete with a frequency response curve that you can click to enlarge. When you move your pointer on top of the frequency response curve, it says it's about to show you "1000b.gif" (the big version of the picture). Now go to the 600 specs page (the 600, not the 603) and look at the frequency response curve. Hmm, looks very similar doesn't it? Let's slide our mouse pointer over it and see where that would take us. When you move your pointer on top of the frequency response curve, it says it's about to show you "1000b.gif" (the big version of the picture). Same damn curve. Is that accurate? Hell no, these two mics sound nothing alike. And that curve doesn't look anything like what either of the mics sound like.

Now let's wander over to the Sound projects page at http://www.pmiaudio.com./ Let's click on the "Specifications" section at the top of the "Studio Projects" page and then let's click on the T3 specs. Curves look interesting, but not exactly what I was expecting, judging from the sound. Let's look at the polar patterns. The upperleft polar pattern is the omni mode obviously, and it's pretty smooth. Let's really study that curve for a minute. Notice the litte variations from perfectly round? Remember those.

Now go back and look at the omni polar pattern for the C3 mic. What's this. According to this these two mics are IDENTICAL in omni mode. Same little variations. So which should you believe? Well let's go look at the C1 for a minute since that's the mic most people are interested in initially. Hmmm, that doesn't look like a cardioid pattern polar response. Wait a minute, I know this curve. I've seen this before - just a minute ago. It's the same omni curve that's being used for the other two microphones.

So are Alan and Brent liars? No, they're not. But the specs sheets are wrong. If you're basing your decision on the specs, you're not getting the right information. I'm not picking on Brent or Alan - most microphone companies don't provide the specs you need to make an honest evaluation of the microphone's performence - and that includes all the big guys too. Speaker manufacturers are just as guilty. I just used Alan and Brent's pages to show you some pretty obvious screwups - some of the other manufacturers' screwups are less obvious ("downright sneaky" would be a better term).

In my next post, I'll get into how they create frequency response curves, different methods of measurements (steady tones, warble tones, white noise, pink noise, 1/3 octave, impulse, etc.), how they control the final results (pen speed, chart speed, smoothed, averaged, etc.), and what the curve REALLY means.

Stick around - it's gonna be fun. You're gonna hear about what manufacturers really DON'T want you to know.

Harvey Gerst :

HomeRec, did I miss pianos? I'll get to it before we finish. I know I wrote part of an article for miking upright pianos for Recording magazine.

Badgas, sounds like a typical blues harp rig. I'd just put a mic in front of your amp and take that signal into the board. Possibly a short plate reverb and that's it. Hard to improve on your basic setup.

It was for Electronic Musician, and here's what I wrote:

Pianos and Mics - No Simple Solutions

Just as with most acoustic stringed instruments, the bulk of the sound is produced by the sounding board to which the strings are attached. In guitars and violins, it's the top of the instrument; in pianos, it's the sounding board. You don't mic the picks, the bows, or the hammers - they produce very little sound.

There are several considerations when placing mics for piano recording. Foremost, will the instrument be recorded by itself, or with other instruments playing at the same time? Those two situations require different mic techniques. Is it a grand piano or an upright piano? Each requires different mic techniques. Finally, where will the recording take place? That may also require different mic techniques.

If the purpose of the recording is accuracy, and you're micing a solo concert grand piano, then you'll need some good, small diaphragm condensor mics, placed some distance from the piano, usually around 6 to 8 feet away. You can use an x-y setup for cardioids, or a wider spaced ORTF setup with omnis or cardioids.

The piano lid is used to direct some of the sound towards the mics. IF the piano is part of a group of instruments, you can get better isolation by micing the underside of the instrument, using a sightly wide spacing with omnis or cardioids. Mics placed inside the top of the instrument can also be used, but it's harder to achieve a good balance or isolation since the piano lid will also reflect sounds from the other instruments into the mics.

Large diaphragm mics can also be used, but the response changes as the sound enters from different angles and the larger mics add coloration (which can sometimes add an unexpected richness to the sound).

Upright pianos should be miked from the back of the instrument, but try to avoid having the soundboard too close to a wall. The distance from the wall will create a standing wave which will interfere with the sound. If the piano has to be near a wall, angle the piano so that it doesn't sit parallel to the wall. Be especially attentive to a ringing sound when micing upright pianos.

This ringing is caused by resonances within the piano, and usually can be solved or reduced by moving the mics around till you find a dead area, free of the ringing. Just as with a concert grand, close micing is not advised, but since an upright piano is usually part of a group, it's not possible to mic from a distance and still have isolation.

To sum it up, first choices for recording a piano would be small omni or cardioid condensor mics, but don't be afraid to try large condensors, ribbons, or dynamic mics (if that's all you have). Mic from a distance if possible. Second choice would be under the piano, and finally, from the top of the piano, but watch out for ringing and reflections from that position.

Wil Davis :

Any thoughts on using a couple of PZMs taped to the inside of the lid of a grand piano?

Harvey Gerst :

Wil, PZMs taped to the the underside of the lid wouldn't exactly be my first choice for miking a grand piano. It might be ok if the piano was being used in a rock setting, but even then, I would probably try some mics under the piano first.

Posted by muzeman on 10-06-2001 03:43:

I read what you said about recording both acoustic guitar and vocals at the same time with three mics. I own 1 good large diaphragm mic and have tried to do this many times with no success. I’m looking now for a small mic or mics to accomplish this.

My question(s) is,

a) I record in a small room and am concerned about having the two mics for the guitar within the proximity of the body,will I get to much boominess(is that a word?)

b) I was looking at either a pair of 012s or a single SM81,because it has a 3 position bass roll off,which I thought might help with this.

c) How much difference do you think would be between stereo and mono for this type of recording?Most of it is going to be mixed with other instruments later,bass drums ect.

d) Do you think an 81 is the best way to go for a single mic? If I go with the 81(or another mic you can recommend) do you think Shure or other manufactors are consistent enough where I can purchase another one later and use them for a stereo pair?

e) And if you or anyone else knows,is Octave the only one who sends their seconds to GC or could I run into problems with other mics bought there?

Harvey Gerst :

a) A pair of omni mics will reduce the boominess, or a figure 8 pattern mic turned sideways will work, with the mic's null pointed at the singer.

b) The MC012s with omni capsules would work well for the guitar if you move them in pretty close to capture just the guitar. but placement will be very critical.

c) In that case, unless the stereo guitar is the most important element of the song, forget about recording it in stereo. Go for maximum separation.

d) The 81 is a pretty damn good mic.

e) They don't get Oktava's "seconds", but they do get the uneven QC that Oktava is known for, and that makes getting an Oktava there a "crap shoot".

muzeman :

a) How is the earthworks stuff. I'm looking at the SR71.

b) How do they compare to the SM81? Do you think I can use just one for mono on an acoustic with vocals on another mic,or do you think I would do better with a large diaphram figure 8 on the acoustic?

c) I record in an unacousticly enhanced room(if you know what I mean) What brand of figure 8 would you recommend? I guess it would have to be a multi pattern?

Harvey Gerst :

a) It's VERY good. That's a nice hypercardioid mic.

b) Very well, but the -10dB pad and the 2 position roll-off filter on the SM-81 might give you a little more flexibility for recording guitar.
The figure 8 pattern will almost completely eliminate the voice from the guitar track. Other than recording them separately to begin with (which would be the best way), a figure 8 pattern on the guitar would be best.

c) You can hang some packing blankets on mic stands to create a smaller, deader space to record in. The Studio Projects C3 and the Nady SCM1000 are both pretty good, low cost, multi-pattern mics.

Harvey Gerst :

Microphone Frequency Response - The Window

Most microphone manufacturers quote frequency response numbers somewhere on their spec page, and it's usually something like "20 - 20 kHz" (or "30 - 15 kHz"), but what does that really mean, and how does it relate to what you hear?

For that you'll need to know how to read a frequency response curve, add in what they "don't tell you", and understand the amount of deviation possible between identical units. But before we can do that, you need to know how microphones are measured.

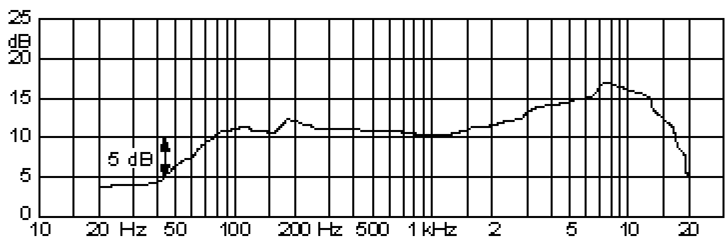
Even though computer measurements have replaced a lot of the mechanical measurement systems, companies (like B&K) still provide precision microphone test equipment, consisting of a frequency sweep oscillator, synched to a chart recorder, and a ruler flat test microphone.

Basically, you feed the oscillator signal into something that will generate the sound, hook up the mic you want to test, and the calibrated mic, then sweep the entire audible frequency range while you chart the "difference" between the calibrated flat mic and the mic you're testing. The resulting chart is the frequency response of *that one microphone*.

Calibration mics usually come in two flavors: direct measurement mics (on-axis), and diffused field mics (usually 90° off-axis). Direct measurement mics are used in anechoic chambers where there is no sound bouncing around so the mic can be designed to be absolutely flat on-axis (i.e. pointed straight at the sound source). As you aim the mic away from the sound source, the high end response of the microphone drops off dramatically.

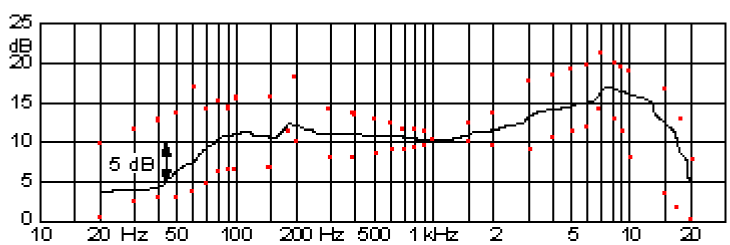
Diffused field microphones are used in normal type rooms where pointing the mic directly at the speaker will pick up unwanted reflections. When making measurements with diffuse field mics, they're usually pointed 90° off-axis (towards the ceiling, the floor, or one of the side walls. Diffuse field microphones are flat 90° off-axis, but they have a large rising frequency response on-axis.

So we now measure our mic, using one of the two methods described above and we look at the chart that was produced, but that only tells us about that one mic. . Here's the mic curve for "our mic":

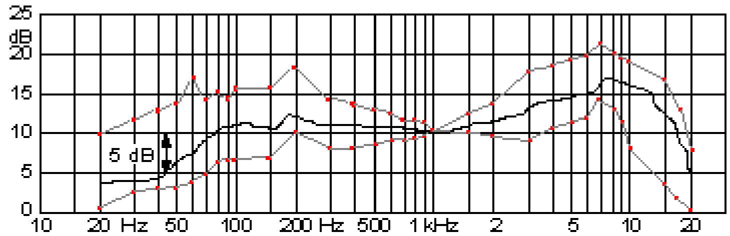


We'll need to run a batch of the same mics to see how much they'll vary from this one mic we just tested. To make it easy to compare the frequency response, we'll adjust the level so that each mic is set to the same level at 1,000 Hz (although we'll keep track of how much the level needed to be adjusted for each mic). Let's say we test 50 mics. We lay out the 50 charts and we also have a blank piece of chart paper in front of us.

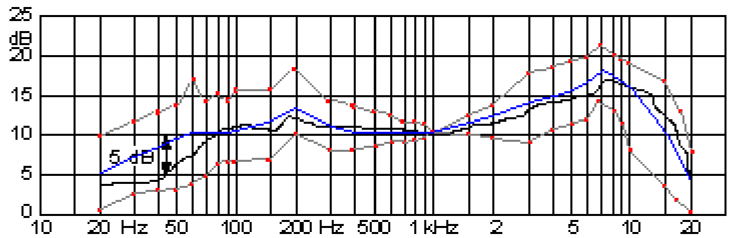
We find the lowest frequency (20 Hz) on each chart, and look for the highest signal level (loudest), and the lowest signal level softest) we measured at 20Hz. We put two marks (shown in red)on our blank piece of chart paper at 20 Hz. We do the same thing at each line, peak or dip on the chart, until we have an upper and lower row of dots that represent the maximum and minimum range of frequency responses from this batch of mics. Here's the curve for "our mic" and the variations we found in testing 50 mics:



We then connect all the upper red dots, and we connect all the lower red dots (with the final curves shown in grey):

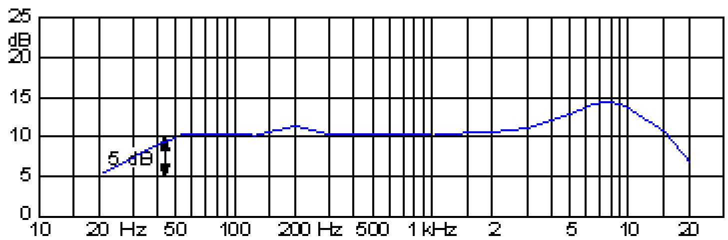


We can then draw a line (the blue curve) exactly centered between the upper and lower dots and that's our "typical response curve" that we submit to the marketing department. Understand, at this point, the curve could look fairly flat, but individual mics can vary by 5dB or more from the "average curve", and still be considered "normal". (Remember we also adjusted the output level for a constant 1,000 Hz signal from each mic? That will thow off the results even more and be critical when it comes to finding a matched pair).



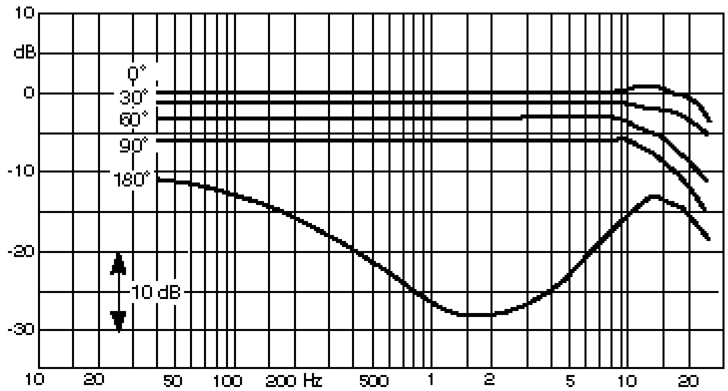
Well, our sample (in black) isn't too far off the average (in blue), but we might find some mics in that batch that are better in the bottom end. How tight to hold the "deviation from average" window is a judgement call by the company and then carried out by the quality control department. At companies like Neumann, they use a 4dB window, which means that all mics must fit within a +/- 2dB window (4 dB overall) of their published curve. B&K test mics may use a window as small as +/- 1/10th of a dB variation from their published curves.

But our "average curve" may still look "too jagged" for public consumption" from the marketing department's point of view, so the curve can be "smoothed" by averaging some of the jagged peaks, or slowing down the pen speed on the chart recorder (so it doesn't move as fast up and down and makes the curve look smoother by simply ignoring all the little jagged short bursts). These are usually marketing decisions, so that "our curve" look similar to "other companies' curves":



And there we have the final "respectable" frequency response curve that is published in the advertizing literature.

Now, here's another "gotcha" for most pressure gradient mics: the frequency response will change, depending on the distance from the sound source, or the angle to the mic. Some manufacturers will actually show the "proximity effect" on the frequency response chart, showing how the bass is boosted as you get closer. Some will also show the frequency response at different angles (usually 0°, 30°, 60°, 90°, and 180°), like this:



When you look at a number like "Frequency Response: 20 - 20k", look at the published curve to see what the "usable response" really is, and remember that the curve you see is "averaged and smoothed. Unless the deviation is shown (either as a gray area or a line above and below the curve, or a number like +/- 3 dB), you really don't know what **your mic** is really doing. That's why it's so hard for the average person to tell what a mic might sound like, judging from the frequency response curve, or just reading the specs.

Any questions so far?

Dobro :

But you can trust a Neumann frequency response chart to be within 4 dB of what they say it is... Hmm. So one thing you're paying for in a Neumann is a mic that is what it claims to be on paper?

You know, I found an EQ preset in Cool Edit Pro that sounds good on my voice tracks - they call it 'Mackie High Band', but it matches the frequency response chart for a TLM 103. I wonder how many of Cool Edit's presets (or those of other editors) are nothing more than the published frequency response of famous mics. I think I'm going to do some tweaking with my Cool Edit EQ.

Harvey Gerst :

You can trust it to be within 2dB of the pubished curve - in each direction. It should never be more than 2 dB louder or softer than the published curve, once you match the levels at 1,000 Hz.

Bingo, that's what some of the simpler mic modellers do.

Three more things I forgot to point out.

1. There is no mic (in any batch you test) that will match the advertised blue curve.

2. If you happen to get a worst case mic that has the horrible peak at 200Hz and at 7,000Hz, it might sound very bloated and screechy if you have a singer with a lot of energy in those ranges, or it could sound "full and detailed" if the singer doesn't have a lot of energy in those areas.

3. Even though the response takes a nose dive after 10kHz, and starts to rolloff below 100Hz, it is still capable of responding to energy from 20Hz - 20kHz, and the manufacturer can advertise it as such (and not bother to publish a curve).

BTW, the first 5 mic graphs shown in the above example were all hand-created by me; no microphones were actually harmed or used during the making of these curves. The very bottom curve (showing off-axis responses) is a real curve of a real DPA mic.

Henrik :

Yeah, I have a question, or rather a clarification so I'm sure I understand this properly:

a) Assume you're comparing some mics of the same model, as in your second jpeg, with the red dots. At 200 hz for example, one mic reaches 10 dB and another mic reaches 18 dB.

Now suppose that apart from the 200 hz deviation, these two mics have similar response curves (a very theoretical assumption no doubt). Would you then be able to make a recording with the mic with the 10 dB response, and on your EQ boost the recording 200 hz 8 dB, and as a result have the sound you would have obtained if you had recorded with the other mic?

b) If so, then I can understand that's a helluva difference! I mean boosting any recording 8 dB really alters it A LOT. Are there any real mics out there that show these big differences between the individual mics? Even Neumann's guaranteed +/- 2dB could really make a noticeable difference (if you were unlucky enough to get hold of two mics on each end of the spectre).

c) Can we trust manufacturers that sell what they claim are matched pairs of their mics?

Posted by Harvey Gerst on 10-09-2001 00:07:

a) Assuming the peak was the only difference and you could match the Q of the peak (the Q is what determines the shape of the frequency boost or cut) with a parametric equalizer, would that eliminate it? Not exactly.

See, that's a resonant peak, which means something in the design is resonating at that frequency. A parametric eq won't make the peak go away entirely because there's always gonna be some resonant energy hanging in there after the note (that excites it) stops.

b) Take a look at the curves from Beyer, Sennheiser, and a few other major companies; +/- 2dB is VERY good as far as tolerances go. To be fair, most of the big names do hold good tolerances but it can get hairy in the high frequency end of the spectrum on the lower priced models.

c) Usually, yes. The manufacturers who sell matched sets can usually use their test curves to find two mics that are similar in response and sensitivity. (Kinda like going thru those 50 curves we ran and finding the two curves that are very similar: that's our "matched pair"of mics.) Not necessarily the "best two" mics, but the "closest two" mics.

dobro :

a) A comment, a question. So, in the end, forget the published graphs and trust your ears.

b) However, the Neumann published graphs are pretty trustworthy, from what you say. Any other companies you know about Harvey, that put out more or less reliable frequency response charts, 'smoothing' notwithstanding?

Harvey Gerst :

a) Well, kind of. You CAN trust some of the curves, once you know how to interpret them, and if you can find the manufacturer's stated tolerance (usually buried in the fine print, or a dB range printed right on the graph itself). When you see a wide tolerance number, ask yourself "Why do they need a large tolerance if their quality control is capable of much smaller numbers?"

b) Shure is pretty accurate, and so are many of the good quality mics (like Schoeps and Earthworks and DPA and B.L.U.E., just to name a few). I'm sure Soundelux and Brauner and Soundfield are also pretty honest. All the Oktavas I bought from the sound room had curves with them that matched very well with what I heard.

Harvey Gerst :

Sensitivity - What's that all about?

Sensitivity is the measurement that tells you how hard your preamp is going to have to work to get the signal up to a useful level. It's found by feeding a specific sound level into the microphone and measuring the output level of the mic.

The older standard was µbars (where 1 µbar equaled a 74dB SPL). The new standard is Pascals (where 1Pa equals a 94dB SPL). If the measurement is shown in µbars, simply add 20 dB to the ouput level to convert it to Pascals. Here are some typical microphone ouput levels:

1.1 mV/1Pa = -59dB (very low output - requires almost 60dB of gain to hit 0 on the meters - typical ribbon mic output)

1.2 mV/1Pa = -57dB

2 mV/1Pa = -54dB (typical dynamic mic output)

2.3 mV/1Pa = -53dB

5.6 mV/1Pa = -45dB

10 mV/1Pa = -40dB (typical condenser mic output)

20 mV/1Pa = -34dB

25 mV/1Pa = -32dB (very hot condenser mic output)

If your preamp gets noisey at high gain, avoid using mics with a big negative dB number. All that -dB number is showing is how much preamp gain you're going to need to bring the signal up to a useful level.

Finally, you may see a number thrown into the sensitivity measurement that says "+/- 1.5dB" or "+/- 2dB" - that's how much variation in output is allowed by the manufacturer between units of the same model of mic. "+/- 1.5dB" means that one mic may have 3 dB more output (or 3 dB less output) than another mic of the same exact model.

Chessparov :

Harvey, what do you consider a good SPL rating for vocal mikes, and have you had any singers "blow out" any?

Harvey Gerst :

A mic in the 128 to 135 dB max SPL level should be more than adaquate for most singers. I've never had anyone blow out one of my vocal mics.

Maximum SPL - How Loud Can You Go?

Since chessparov just brought it up, let's discuss "Maximum SPL"and what that specification means.

"Maximum SPL" is the maximum Sound Pressure Level a microphone can take, at a specified level of permissable distortion.

The problem with this spec is that some microphone manufacturers don't tell you what the distortion level is, or whether it's the capsule or the mic preamp (inside the mic body) that's distorting, or they calculate the level at a distortion level that's different from other manufacturers. If the distortion isn't mentioned, figure it's either 1/2% to 1% (tolerable), or 5% (starting to get gross).

If they show only one distortion vs. SPL figure, it's easy to convert that number to distortion figures other manufacturers use, to help make a fair comparison. Here's how:

For a round diaphragm mic, distortion will usually double for every 6dB increase in SPL. So, if someone shows a max SPL of 1/2% @ 128dB, it's gonna be around 1% @ 134dB, 2% @ 140dB, and around 5% @ about 148dB.

You can control too much output level by two methods: by placing the artist further back from the mic (which will also help reduce wild variations in level due to movement), or by using compression to smooth out level inconsistancies, but with the liability of a possible increase in room noise.

As you move the person back, the inconsistancies from note to note smooth out, but you pay the price of added room noise.

Usually, you'll want to strike a balance by moving the artist back just a little (to control levels, but not so much that you pick up a lot of room sound). For condenser mics, I like somewhere between 1 and 2" away (for ballads and very soft singers), and 6 to 12" back (for the "belters"). I might also add some compression if their levels get really wild (anywhere from 2:1 to 10:1 ratios, but only triggered on the VERY LOUDEST peaks).

The final section (on polar patterns) is coming up next, and that should wrap this whole thread up.

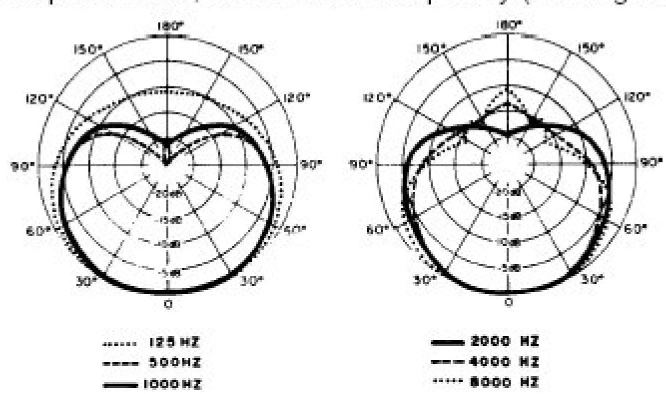
I'm sure they'll be some questions, and I'll try to answer them all, if I can. Thanks to everybody for hanging around this long.

Polar Response - Turn, Turn, Turn

Ok, I noticed my last post was #299 in this thread, so I might as well finish this whole thread off at post #300, so here's the last section, on polar response. I'm gonna use the sheet for the Shure SM-57 as the example, so download this and follow along:

Polar Pattern

Unidirectional (cardioid), rotationally symmetrical about microphone axis, uniform with frequency (see Figure 3)



TYPICAL POLAR PATTERNS
FIGURE 3

http://www.shure.com/pdf/userguides/guides_wiredmics/sm57_en.pdf

Notice the polar response looks very smooth and you can almost visualize what it would sound like if you moved 30° off to the side of the mic, but what "you see" isn't exactly what "you get". Look at the frequency response curve to see why. Notice that the response of the mic is down a couple of dB at 125 Hz and it's got a 7 dB peak at around 5kHz or so?

Now look at the polar response curve and find those 2 frequencies. Notice at 0°, they show up on the 0dB line? If you were on axis, the 5kHz signal would actually be 7 dB louder than a 1kHz signal, but they're shown as identical levels at 0°. In other words, they've been "normalized" to be the same level at 0° as all the other frequencies. In reality, that 5kHZ line should be 7dB louder on the graph, all the way around.

The only thing the polar pattern shows is the general pattern of the mic at different angles; it does not represent reality with regards to the actual signal levels you may get off axis. For that, you have to use the frequency response curve and extrapolate (i.e., "guess") the actual off axis response from there.

And that's the "last secret" of this whole thread. We started off discussing "diaphagm size", and this last post covers the polar pattern part of the question, although we discussed different polar patterns and their use earlier.

I hope you enjoyed this as much as I did, and I'll try to answer all questions, and clarify anything I din't explain as well as I should of. It's a big subject and I didn't go into it as fully as I maybe should have, but tried to make it as useful as I could to the broadest spectrum of posters here. If it was too simple and basic for some of you, I apologize, but just look at it as a refresher course. For the rest, I hope it's given you some insights into how to improve your own recordings. Whatever - I've enjoyed the hell out of this.

Sincerely,

Harvey Gerst,

Just An "Old Fart" Recording Engineer

PART III (conversation)

chesterfield :

Harvey......... with regard to close-micing loud sources (guitar amp, snare, trumpet), isn't high SPL the most important factor? I just bought a Shure SM57 because it's universally acclaimed as THE mic for those applications. Is it because this mic has high SPL?

It's been said before, but thanks again for all your time and effort putting this info together. Now I've got to go back and find the link to print out the main text so I can save it and refer to it often.

Harvey Gerst :

Actually, it's the fact that the frequency response of the mic complimemnts the sound of the snare and electric guitar that makes it so desirable. It adds a nice high frequency boost at the top end and has a natural rolloff at the bottom end, just damn near perfect for rock snare and electric guitar.

Henrik :

Hmm...I used to think SPL meant that louder sounds might break the mic (so a low SPL figure meant something like "guarantee is void if you stick this into a kick drum").

One thing I wonder: Is it always desirable to have as high SPL as possible? Or are there situations where it could be an advatage to use a mic with a lower SPL?

Harvey Gerst :

SPLs way louder than the maximum SPL usually means increased distortion. Some mic designs (like the Sennheiser MD421) can handle 150dB without any problems - so can a lot of small omnis.

In general, large diaphragm condenser mics CAN handle quite a bit, but the internal preamp often overloads first, so a pad is used to lower the capsule outputby 10 to 20dB typically. Kick drum is a special situation, since the main concern is the draft of air the kick head causes. That blast of air is what can kill the mic, not the high SPL level.

Some mic designs ((older ribbon mics for example) CAN also be damaged by high SPLs or drafts from fans, speaker ports, doors closing, or even shutting the lid too fast on the carrying case.

So why use a ribbon mic (that typically has a low SPL rating)? Ahhh, the sound is wonderful. Many ribbon mics use ribbons that are only 7/10ths of a micron thick (a typical human hair is about 20 microns thick by comparison).

The ribbon in a ribbon mic has essentially minimal mass and responds beautifully to a lot of signals, such as voice, strings, horns, etc.. The resonance can be set very low (20Hz on an RCA 44BX), and that single piece of corregated aluminum ribbon has almost no other resonances, so it's very flat and smooth throughout its entire response range.

An RCA 44BX or 77DX on a Marshall 4x12 cab, about 2 feet away, is a sound to die for. Lush, rich, beefy, gorgeous, you pick the adjectives. As the frequency goes up, the SPL rating of most ribbons increase dramatically - about 6dB more power handling per octave. MAny pros will use a ribbom mic or large condenser mic for kick, by placing the mic away from the kick (4 to 6 feet), in a long packing blanket tunnel in front of the kick.

cyork :

Are there frequency bands that tend to distort first? Are various mics sensitive to distort over specific frequency bands depending on their internal resonances? Proximity effect comes to mind as an obvious low frequency problem, but is it always in the low and low-mid that the mic will hit its max SPL first?

Harvey Gerst :

Well, the diaphragm has to move the furthest at low frequencies, so that's usually where the problem starts. For the same output level, a mic diaphragm has to move twice as far for each lower octave.

HomeRec :

That said, I could just be missing something, but where is the maximum SPL listed for the SM57? Do most manufacturers advertise this specification? If not, is there a way to figure it out?

Harvey Gerst :

You're right, Shure doesn't show max SPLs for the SM-57, 58, the Beta versions, or the SM-7. I'd guess at around 138 to 145dB max SPL for a good quality dynamic, but that's just a guess.

vox :

Harvey, the MXL 603 specs claim 137db for 0.5%THD, I know that you have a pair and that you have stated that some of their printed specs may not be completely accurate, do you think this SPL rating is correct? would you feel comfortable using the 603 on a snare or cranked up amp for example?

Harvey Gerst :

That 603 spec sounds about right. For 1% distortion, that would translate to about 142dB. I wouldn't be worried in the least about using it on a snare or a loud guitar cabinet.

HomeRec :

Apparently, this is Shure's answer to "What is the maximum SPL rating for an SM57?"

Can a dynamic microphone handle really loud sounds?

Question

What is the maximum sound pressure level that a dynamic microphone can handle without distortion?

Answer

Realistic Maximum Sound Pressure Levels for Dynamic Microphones

Microphone users often ask "What is the maximum sound pressure level that a dynamic microphone can handle without distortion?" Using the Shure SM58 as an example of a typical dynamic microphone, Shure Engineering performed experiments to answer this question. Like most technical matters, the answer is not simple.

As a point of reference, 140 dB SPL is the accepted threshold of pain for the human ear. The maximum sound pressure level (max SPL) from a human voice as measured by Shure is 135 dB SPL at 1 inch from the mouth. A kick drum played very loudly may exceed 140 dB SPL, but has never been measured by Shure above 150 dB SPL. The loudest orchestral instrument, a trumpet, can theoretically produce a MAX SPL of 155 dB SPL at 1 inch, but only in its upper register. Note that the distribution of energy (sound pressure) in speech, music, and noise is dependent on the frequency. For example, the human voice does not produce much energy below 100 Hz and its frequency of MAX SPL would be higher than 100 Hz. Exactly how much higher depends on the individual voice.

Unlike a condenser microphone which has internal electronics that may overload, a dynamic microphone distorts when its diaphragm hits a physical barrier, like the magnetic pole piece, and can move no further. The excursion of the diaphragm is frequency dependent and the excursion is greatest at the resonant frequency of the diaphragm. Therefore, the MAX SPL of a dynamic microphone like the SM58 is frequency dependent. This means that low frequencies will produce distortion at a lower SPL than higher frequencies.

For the SM58, the frequency range to first exhibit distortion is centered around 100 Hz, close to the resonant frequency of the microphone's diaphragm. At 100 Hz, the measured MAX SPL is 150 dB SPL and the electrical output of the microphone is 0 dB V or 1.0 volts. Note this is a line level signal, not a mic level signal.

In the 1 kHz range, the SM58 measured MAX SPL is about 160 dB SPL due to the change in microphone sensitivity at the higher frequencies. The electrical output of the microphone at 160 dB SPL is +10 dBV or 3.2 volts.

In the 10 kHz range, 180 dB SPL is the MAX SPL of the SM58. However, this is a calculated measurement as Shure Engineering had no means to create such enormous and dangerous SPL. For comparison, NASA reports that a space shuttle launch measures 180 dB SPL and higher at 10 meters.

In the 20 kHz range, the MAX SPL is calculated to be around 190, due to the response falloff of the SM58. But now the point of absurdity has been reached because at 194 dB SPL the sound pressure varies from twice normal atmospheric pressure (at the wave peak) to a total vacuum (at the wave trough). Plus the sound source must be moving at the speed of sound just to generate a wave of this intensity.

In summary, a well-designed dynamic microphone of professional quality will never reach its distortion point in "normal" conditions. If one does encounter distortion when using a professional dynamic microphone for an extremely loud source, it is most likely that the electrical output of the microphone is clipping the input of the microphone preamplifier. [Remember that at 150 dB SPL, the SM58 will provide a line level output!] To solve this problem, an in-line attenuator ("pad") must be placed before the preamplifier input, or the microphone must be moved farther from the sound source. In general, the sound pressure level will decrease 6 dB for each doubling of the distance.

Harvey Gerst :

I think my answer is pretty consistent with Shure's answer, and I think my numbers may even be a little more realistic, since I believe they're talking about what kind of SPLs would cause physical damage to the mic (as opposed to moderate distortion which I was discussing). We both talked about how the low end is the most damaging.

HomeRec :

So the maximum SPL rating for an SM57 would vary according to the kind of sound being recorded, instead of there being a single SPL rating for all applications?

Harvey Gerst :

Well, actually, kinda, sorta, yeah. It depends on the mic design, the resonance of the diaphragm, and the day of the month, but in general, max SPLs are often measured at either 1 kHz, or around 250 Hz. Ribbon mics, like the Coles 4038, even specify the maximum permited SPL at different frequencies.

Most mics can handle most signals, but ribbon mics and a lot of pressure gradient condenser mics don't like singers in close, singing F, P, B, V, W, and T, or anything that produces an air blast, which can bottom out the diaphragm or stretch the ribbon.

muzeman :

I was thinking,watching groups performing on T.V. they always use a dynamic or black electret for vocals and a small condenser for acoustic. The sound they get is equal to or better than the studio sound. (Stained was on unplugged the other nite,they were fantastic!)

Do you think this is a possible way to go in a home studio? It seems like the plus for a dynamic or black electric on vocals would be elimination of room and backround noise, and the advantage of good close proximity effect for untrained voices,one of which I have the pleasure of owning. The big disadvantage being,you don't get the sound of a large diaphram. Am I on the right track with this?

I'm also a little confused about the different cardioid patterns, can you clarify the differance between cardioid,hypercardioid,and supercardioid. Are there advantages and disadvantages to the different designs?

Iv'e been looking at the Shure beta 87a and c,one is cardioid and the other is supercardioid,both black electret I believe.

Also the sm7,which is a dynamic. I'm not sure which would be the best for my situation,any advice or other mic options would be appreciated.

Harvey Gerst :

It's a little tricky to explain without the design theory, but lemme see if this'll help a little bit: Cardioids have a heartshaped polar pattern at most frequencies, but they tend to be more omnidirectional at low frequencies.

Hypercardioids are less wide compared to cardioids, but still have some omni characteristics at lower frequencies.

Supercardioids are similar to hypercardioids at high frequencies, but they act more even at low frequencies by creating a deeper rejection point at around 125° off axis.

So what the hell does all this mean when it comes to choosing the right mic, based on polar patterns?

If you're getting a mic for recording just your voice, it's easier to use a smooth cardioid mic that will be fairly flat and natural and you don't worry too much about picking up bleed from other instruments due to the wide pattern of most cardioid mics.

If you're playing guitar at the same time, you want to try and keep the sound of the guitar out of the vocal mic, so you need a tighter mic pattern (like a hypercardioid) and you try to put the guitar in the null of the pattern so that it doesn't get heard by the vocal mic. And it holds true for the guitar as well;

you might use a second hypercardioid on the guitar to keep the vocal out of the guitar mic. But hypercardioids aren't perfect, especially at lower frequencies.

That's where the supercardioid comes in; it's got a solid null point at 125° at just about all frequencies.

So why not just use hypercardioids and supercardioids for everything? Part of the problem is that hypercardioids and supercardioids don't always have the best frequency response, so you pay a price in performance for that deeper rejection. And they have more proximity effect (which is not always a good thing).

For stage and live work, the rejection in a hypercardioid and supercardioid mic is a blessing, especially when working with on-stage monitors, but it's not as important in the studio, where accuracy counts more.

Does it make a little more sense now?

BTW, it's a "back electret", not a "black electret". That means the condensor element is pre-polarized (carries a permanent charge) by putting the charge on the back plate rather than the diaphragm. The difference in voltage between the diaphragm and the back plate is how a condenser mic works.

gershwin :

a cappella

Harvey, what about recording a 4 part male a cappella group? We have 13 guys, 4 basses, 3 baritones, 4 Tenor 2's, and 2 Tenor 1's. At any one time, we have 11 guys singing backgrounds, 1 soloist, and one Vocal Percussionist.

What would you suggest as far as recording this group? Would it be best to record the entire group together, further out, with an X-Y? Should we add to that mics for each section? Or should we record each section separately?

Harvey Gerst :

Wow, I'd love to record you guys. Without hearing you or seeing your staging, that's a difficult question to answer. If I just had two mics, I'd probably start with X/Y or an NOS pair and adjust the spacing till the middle sounded right to me. If I were to add 2 mics, it would be for the soloist and the vocal percussionist next. Finally (assuming 8 tracks), four more mics to spot each section.

Two mics are all you need usually, but you really hafta get the angle and distance just right to get the full stereo spread, and the right balance of natural reverb. I'd get a "gofer" to move the mics around while I listened to the sound. looking for the "sweet spot" where it all comes together.

muzeman :

Hi, I was wondering if Harvey or anyone else has used the Neumann KMS105,I have a chance to get a good deal on one. Thinking about trying it on vocals,with another mic on acoustic to record both at the same time.

Harvey Gerst :

Every engineer whi has used the the Neumann KMS105 has raved about it. These comments are from people I know who's ears I trust. They say it's the finest stage mic they've ever heard.

muzeman :

Harvey, Thanks once again for your input,I think I'll give it a try. I havn't heard anything bad about it,and if it dosn't work out I'll have a great stage mic and I can start saving for a good figure 8!

Eurythmic :

I'm a home recordist on an extremely tight budget (aren't we all?). I've used nothing but a Shure SM-57 to record all of my tracks, since the very beginning. I'm getting to the point though, where I really hate the way my vocals sound with the mic. I've been working for about six months on my first EP, but with the vocals I've really done little more than record scratch takes for each song - I've been putting off the final vocals for as long as possible, because I'm just dreading the point where I'm going to have to record my vocals and actually *keep* them.

Erm, I'm *hoping* it's the mic, anyway, and not me. I'm not the best singer in the world, but I also know that there have been a lot of vocalists with poorer technique than mine, whose vocals sound far better on record. I guess part of me is imagining this "perfect" microphone that would actually make recording vocals fun again, kind of like the hallowed old acoustic gutiars that seem to make music when you just touch a fret.

But I don't really have a wide knowledge of microphones - I just picked the SM-57 because everyone told me to. That was a good three or four years ago. Now I understand WHAT makes it good, but I don't have a varied knowledge of what else is out there. For a guy like me who can't really say, "Oh, mic x? Well, that sounds very y, and really z, and a, b, and c are good examples or recordings that use it," how am I supposed to know what to look for in a new microphone? I could possibly at least find a store with a liberal return policy if I just knew where to begin. For instance, would it be possible to listen to a recording made with an industry standard mic (such as the SM-57) and say, "You know, you might give this mic a shot. I think it might suit the particular character of your voice" - or does it not really work that way?

My other question is pretty random. You may not know the answer, but you seem to be a veritable fountain of random information, so I'll take my chances. I realize that there aren't a lot of people left who still like this band, but Duran Duran are one of my biggest influences. Would you happen to know what vocal mic Simon LeBon was using during the "Rio" era? I'm thinking it must be the same microphone across the whole album, because that effect is there on every song. Although I'm not enthralled with the sound on that album as a whole, I really like the way the vocals sound. I wouldn't want to sound like that all the time, but if I could figure it out, it would be a great new color to add to my "arsenal" (which is currently sadly lacking in variety).

Posted by nezpierce on 11-27-2001 19:57:

Search Colin Thurston...

Colin Thurston was the producer for the first 2 Duran Duran records, along with virtually every other huge english act around the late 70's early 80's.

Based on that pre-digital effects era, you are going to be dealing with high end analog equipment and engineers who knew how to use a room/studio to get a good sound and probably alot of those "bowie" tricks...

Harvey Gerst:

Duran Duran's "Rio" was done at Sir George Martin's Air Studios, London. You could go to http://airstudios.com/ and see if someone there has the tracking session sheets. If that doesn't work, here's a list of the mics they have (and it's a safe bet they probably used one of these):

AKG
1 CK5 (capsule for 451)
2 C12VR (reissue of classic valve)

9 C414 (large diaphragm condenser)
3 C460 (condenser)
8 D451 (cardiod condenser)
2 D12 (cardiod dymanic)
2 D112 (cardiod dymanic "the egg")
1 D19C (cardiod dynamic)
8 D190E (cardiod dynamic)
2 D202ES (cardiod dynamic)
1 D224ES (cardiod dynamic)
2 D25 (cardiod dynamic)

Beyer
2 M160 (hyper cardiod double ribbon)
2 M201 (hyper cardiod dynamic)
2 M260N (hyper cardiod ribbon)

Blue
4 Blue Bottle (multi-capsule tube mic system)
4 B4 Capsule (perspex sphere pressure omni)

Coles
11 4038 (classic BBC ribbon)

B+K
2 4003 (omni condenser)
2 4007 (omni condenser)
2 4011 (cardiod condenser)

Crown
2 PZM (pressure zone effect)

EV
1 RE15 (cardiod dynamic)
3 RE20 (cardiod dynamic)
1 PL80 (cardiod dynamic)

Manley
2 Valve

Nakamichi
1 DM1000

Neumann
6 U47 fet (large diaphragm)
3 M49 (classic large diaphragm valve)
3 M50 (classic spherical omni valve)
3 M150 (spherical omni M50 reissue)
4 TLM50 (spherical omni)
5 U67 (classic large diaphragm valve)
2 KM83 (minature omni fet 80 series)
9 KM84 (minature cardiod fet 80 series)
4 KM86 (minature selectable fet 80 series)
1 KM184 (minature cardiod fet 180 series)
14 U87 (large diaphragm selectable pattern)
6 U87Ai (large diaphragm selectable pattern)
2 U89 (large diaphragm 5 polar patterns)
6 TLM103 (large diaphragm cardiod)
5 M147 (large diaghragm cardiod tube)
5 TLM170 (large diaphragm fet 100 series)

Pearl
1 DC21 ()

Reslo
1 RV ("the bullet")

Sanken
2 CU41 (cardiod condenser)

Schoeps
12 CMC5 (condensor microphone body)
6 CMC6 (condensor microphone body)
3 MK2H (omni capsule)
3 MK2S (omni capsule)
7 MK4 (cardiod capsule)
9 MK21 (wide cardiod capsule)
2 MK21H (wide cardiod capsule)
3 MK41 (hyper-cardiod capsule)
1 Schoeps Stereo (Dual Capsule Microphone)

Sennheiser
14 421 (cardiod dynamic tom mic)
3 441 (similar to 421)
2 MKH20 (omni condenser)
6 MKH40 (cardiod condenser)
2 MKH50 (hyper-cardiod condenser)
4 MKH800 (extended response switchable condenser)

Shure
10 SM57 (classic instrument cardiod dynamic)
1 SM58 (classic vocal cardiod dynamic)
1 SM98 (minature super cardiod condenser)

Sony
3 C8000 (modern valve)

If I had to bet money on which mic he used (without any more information), I'd bet Simon used one of their 5 Neumann U-67s.

Henrik :

I was wondering, suppose you have a pair of subcardioids like the 603:s (which I don't since Marshall lack distribution up here in the cold North). Anyway. Suppose you'd like their polar pattern to be a bit more narrow, for instance if you want to x/y mic with them, or if you are trying record a guitar while the guitar player is singing into another mic at the same time. Could it then be a good idea to tape some foam rubber directly onto the mic, forming a small tunnel around the diaphragm, but of course leaving the front opening uncovered?

I've seen some mics that come with an optional pipe, about a foot long, which you can attach to the mic in order to make it very directional. ALso the AKG c1000 comes with a small plastic cover that you can put over the diaphragm to change the polar pattern from cardioid to hyper. I was just thinking that this idea may be applicable to other mics. What do you think?

Harvey Gerst :

If you block the side vents of a cardioid, you get .... an omni mic. Ever wonder why a singer gets feedback when he/she cups the mic? Now you know.

Adding mechanical stuff can work, but it's really hit and miss without proper equipment to see what you're doing to the response. Chances are you'l get some resonant peaks that are more directional.

Henrik :

AH, of course! I should have been able to understand that by myself after reading this thread. A cardiod with side vents covered makes an omni. I'll write it a hundred times.

Folkcafe :

I know this is going to be off the subject but the section on drums prompted this question that I've never been able to get a satisfactory answer to. Do smaller drums record better than larger dia. ones. Part of this is due to alway having drums in the studio that are of the more traditional (larger size). I continue to hear and read this but would like some insight to this issue.

I have a small studio doing the light contemporary Folk sort of stuff. I have a great in house drummer (nephew) but he also plays out very actively. That means his kit travels a lot and takes a beating. I would like to purchase a kit for the studio and have been having trouble pulling the trigger partialy due to this one question. Drum kits are a hard one to "try before you buy" kind of purchase. At least for recording.

I figure having a studio kit would make getting reproducable results easier. We all have our areas of expertise. Mine is electronics and not percussion. Can you offer any insight to this issue?

Harvey Gerst :

If the drums are tuned correctly, you don't need a big set, even for heavy metal. Our most popular kit here at the studio consists of:

a 22"x16" Tama Rockstar DX Kick drum
a 10"x10" Tama Rockstar DX High Tom
a 12"x11" Tama Rockstar DX Mid Tom
a 13"x12" Tama Rockstar DX Low Tom
and a 16"x16" Tama Rockstar DX Floor Tom

The 13"x 12" doubles as a floor tom (replacing the 16"x16") or as the third rack tom, depending on the group.

When tuned and miked correctly, the sound is huge!!

nezpierce :

I would totally agree with Harvey on the tuning part. New heads for the drums (which can get costly if you have to change them every time you record) help immensely if you are going for a real tone-y sound. If you just like the drums to be pitched "thuds", you can get away with older (less than 6 months) or thicker heads (ala remo pinstripes etc).

I find that smaller thinner cymbals tend to work better as well. for example, I once did cymbal overdubs for a record that had a lot of triggered drum sounds and I used a 12 UFIP splash as a crash cymbal. It sounded fantastic!

Folkcafe :

Thanks for the info. Of the 2 kits under consideration I'm thinking of an even smaller Kick 18 X 14. The rest are of similar size as the set listed above.

If I need a larger kit then I can always deal with my nephew's or the clients but it has been a real pain to stay on top of . As a drummer in a hard core rock band my nephew beats the snot out of them and head life is very short. Important sessions always require a half day's work getting the kit right including changing heads. Other work is hit or miss (mostly miss) but it's not critical. Storage space and money is also a problem so I can't buy more than one kit. I have enough trouble with space for the guitar collection.

My studio is just a kind of scratch pad kind of place where clients come to work out ideas and do demo kind of stuff. Then it's off to a "real" studio if the material warrants it. The rest of the work I do is live sound and recording. I have a little niche market that is starting to work out. Covering all the requirements of this kind of gig has made it easier to find work. Anyone who has ever had to balance front of house and monitors knows trying to get them sounding decent and not adversly affecting the recording can be a real challange. This can be really hard when you have to interface 2 separate systems and companies (recording and PA).

The studio is a part-time gig. I'm a full time hardware geek in a corporate studio. I get to play with all the big toys at work but I really enjoy the work I do in my own little place more.

I haven't recorded any dulcimers in the studio but have dealt with them live along with anything else you can imagine. One of my current clients plays acoustic and electric Cello. The instrument is played through a delay and phrase sampler. It is a very wild, unique style. His web site is at: Cellobop.com

I like to work with odd and strange kinds of off beat stuff. If it's really strange then it's for me. Otherwise it's the general singer/songwriter genre stuff.

Henrik :

A question here for you Harvey: I'm in the business of getting a stereo pair of mics for drum overheads etc. The Oktava M012 is a bit out of my price range, but I'll of course go there if I must. But in your big test of Marshall mics (another near classic Harvey thread) you mentioned that the MXL603's sound almost identical to the Oktavas. One difference however seems to be the polar pattern, the 603's are wide cardioid ("near omni" I think you said). When stereo micing, I prefer the x/y setup, since a lot of my stuff goes on the Internet, and mono compatibility is necessary for lightweight files.

Bla bla bla, anyway - do you think the 603's wide cardioid pattern makes them unsuitable for x/y micing?

Plus, do you think it's OK to buy a pair without having heard them, or should they be matched? This would present a problem for me, since I will have to buy them online.

I'm also considering the ADK A51sc - small diaphragm condensator. Apparently it's voiced like a KM84. Tried them? Heard of anyone who has? Any good?

Harvey Gerst :

I would have no problems buying two 603Ss and not having them matched, of course I would like them exactly matched if I could get them that way. I would also pict up a pair of those dirt cheap Behringer reference mics and try those as well, wide spaced.

The 603S should work fine as an X/Y pair. They're wide, but still very cardioid.

I haven't heard the ADK mic.

Middleman :

Picturing the mix

I can't help but wonder as to your thought process in building the final mix. Now I know there would be lots of variables so let me try a scenario, let's say a ballad. Piano, Drums, Acoustic Guitar and vocal, maybe some strings via synth or real and steel guitar.

How do you approach the placement of these instruments in the sound spectrum. What EQ considerations are you thinking about as you mike these and how do you go about building the framework of the core mix?

Do you have a preset sound picture in you mind before you start or do you just mike each instrument to its most natural sound and try to match it up from the board?

Finally, I would like to hear your thoughts on reverb, delay or just ambiance in general that you would use in the scenario above.

Maybe we could take this to a new thread. Any suggestions you could provide would be appreciated.

Harvey Gerst :

The only thing that saddens me about this thread is that the Behringer thread that I started (about cheap $35 mics) has had almost as many views and posts as this thread, which represents years of experiment and experience. There is stuff in this thread that I've never seen anywhere else, in print, or on the net.

This thread developed over months of trying to cover as many aspects of microphone design and technique as humanly possible, while the other thread was "hey, I just noticed this and it might be a pretty good deal".

A little disappointing to me.

Posted by harley96 on 12-28-2001 20:03:

Psychiatrist in need!!!

The saddest part for me Harvey is I've read both threads from the beginning and have re-visited them several times and I still have not bought a pair of ecm's yet.I came close once, I went to 8th street.com clicked on the buy it ordered two of them got my total then "what the hell am I doing" corner of my cranium took over.These mics are like $80.00/pair so certainly affordable but when does my gear addiction end!! I wake up to see recording magazines strewn across my bedroom floor.I take a shit there's my latest mix magazine on the back of the toilet,everytime I get on line my hand clicks on home recording bbs. without thinking about it, my wife sees me at 3:00 am after I've posted my last thread, wake up and do it all over again.I'm tellin ya I can't afford drugs, I even quit drinking 6 months ago, my second biggest addiction (cigarettes) I'm gonna have to give up because the biggest addiction in my life GEAR is consuming everything. Next on my list is trying to get my room acoustics in check!!Never ends!!!

Harvey Gerst:

I think some of the new people to the group might want to look at this thread, which will answer a lot of their questions.

ausrock :

BTW, off topic but I believe you were "involved" with Acoustic amps at one time...........true or false?

Harvey Gerst :

I designed the 260, 360, 150, 140, and 130 series (although there were several other people there that made it a reality), and the 270/370 stuff, etc. I was the Vice President of Acoustic for a few years out of my life. I watched it grow from just myself and Steve Marks to about 300 people at its peak.

PinkStrat :

Wow! I vividly remember when I was a lad of 15 and happened to cross paths with Shel Horlick who was, at that time [1970] a sales rep for Acoustic and ARP [not to mention a few other things that were new and hip] in Northern California. I wonder what Shel is doing now? The last time I saw him was in the early eighties at his house in the Hollywood Hills. He was with Schecter Guitar Research if I recall correctly. Harvey, did you have anything to do with those Delta amps that David Pack used [in the early days of Ambrosia]? This brings back a TON of Los Angeles memories for me... Man, what a long strange trip it's been!

Harvey Gerst :

I was President of Delta Products and those were my amp design. I found out Shel Horlick died a few years ago. His house was near mine (I lived near the top of Laurel Canyon on the Hollywood side, and Shel lived on the Studio City side).

Jeff Baxter brought in Dave Schecter and for a while. We loaned Dave some space at Delta to work on his pickup designs.

Al Sim :

Behringer ECM8000 in X/Y

Harvey, would a pair of Behringer ECM8000s be good for X/Y recording an acoustic guitar that has a lot of bass response? Or would Marshall MXL 603s be better? Any other budget suggestions? The guitar is a wonderful Gibson from the 40's and I'm doing a terrible job with a Rode NT1 and an A-T MB4000C.

Harvey Gerst:

I don't think you'll get enough stereo separation with two ECM8000s in X/Y, but hey, try it. Try NOS or one of the other near coincidenct spacings to see if that gives you a better image.

The use of omnis will certainly help tame the bottom end on an old Gibson (is it an old Gibson J45?). You might also try playing around with different string gauges on the low E and A strings to control levels.

Posted by Al Sim on 01-09-2002 17:17:

The Old Gibson

Yes, I believe it is a J45. The guitar belongs to the singer-songwriter I work with. I don't think I'll have much luck asking him to change string guages, so that'll have to be a last resort. It's a truly great-sounding guitar. I haven't bought the ECM8000s yet. I'm trying to figure out what to get on a budget of $200 for recording this particular instrument. A pair of MXL603s and an ECM8000? Any other mics you'd suggest?

Have you ever encountered an EV 676 and is it good for anything?

Harvey Gerst :

The J-45 can be a very difficult instrument to record. What the audience hears is NOT what the player hears. They can sound downright "tubby" from out in front of the guitar. I'd probably try the ECM8000s first, doing the "over the shoulder" routine, and then dink around with the placement from there.

This is the one guitar where you might need to add some eq, if nothing else works (some "bass" to "low mid" cut, anywhere from 80Hz to 250 Hz,depending on the guitar and the song).

For the right song, there's nothing that comes close to a J-45, but they can be a flat out bitch to record. It's also a lot easier to record a strummed J-45 than it is to record a fingerpicked J-45.

I've had pretty good luck using a Shure SM-81, but that puts it out of your price range.

I'm not familiar with the EV, but most EVs always have "some" use in a studio. Does EV have a historical database on line? You might start there, or ask them via email if they have any information about the 676

Al Sim :

There are songs that I would like to try X/Y on, if not on the J45, then on one of our other guitars. Would a pair of MXL603s be good for that, or is the another mic in that price range that you would recommend? And do I have any hope of getting good results doing X/Y on the J45, or should I not waste time trying? That should be it for questions from me, at least until the mics arrive. Meanwhile, I'm off to the EV website to see what I can learn about the 676.

Harvey Gerst :

Yes, a pair of 603S's in X/Y should sound fine, as long as you're not too close. As far as if it's worth trying, yes - at least you'll learn something about the technique, even if it doesn't work for this particular application (and it might work perfectly for this).

Please keep in mind that even though stereo is great, it's not always the best technique for maximum emotional impact. Just as black and white is sometimes better for conveying a mood (instead of color) in movies, don't be afraid to think in terms of mono instead of stereo when it might be more appropriate.

Henrik :

a) Harvey, I'm thinking of getting some figure 8 pattern mic, and my question is: Will all figure 8 mics give an equal frequency response on both sides of the mic, or can this differ?

b) If so, are mics with selectable polar pattern more likely to have different frequency responses in figure 8 position, compared to a mic that can only do figure 8 (like a ribbon)?

c) I also wonder, for M/S-recording, is it necessary that the two mics have to be very similar (like in X/Y-recording), or can they differ some? Could you, for example, use a C3 for side and an NTK for mid?

d) Any recommendations on good figure 8 mics?

Harvey Gerst :

a) No, it can differ greatly depending on how the figure 8 pattern is achieved. Ribbon mics that are designed strictly as figure 8 mics have the most perfect figure 8 pattern AND the flattest off-axis response.

b) It depends on how well each diaphragm is matched and balanced.

c) Yes, you could use that combination.

d) RCA 44BX, Royer, Coles 4038, Beyer 130?, etc.

Henrik :

But I wonder - you are only mentioning ribbon mics when I asked for some examples of decent figure 8 mics. Do you mean to say there are no condensors with selectable patterns that can produce a decent figure 8?

Also, I understand your favourite method of overhead drum micing is a spaced pair of omnis. Does this mean you're not too worried about the mono compatibility, or do you have a trick up your sleeve in order to make the spaced pair work flawlessly in mono? (Hopefully some of the music you are recording gets aired in radio and TV). Why don't you use the MS technique, which I understand is foolproof for mono?

Just came to think of another thing I never really grasped about the MS-technique (even after having read Wes Dooley's web site. I must be dim. I'm a bass player, OK?):

If I undstand it correctly the sound of the mid mic is subtracted to the sound of the side mic on one channel, and added to the sound of the side mic on the other channel. But what exactly does it mean to subtract a sound from another (as opposed to adding one)?

Harvey Gerst :

Gimme another day to get over this damn flu and think about it for a little bit more. Yes, you can use almost any figure 8 mic as the "side" mic, but there are some other considerations. Lemme see if I can make this easier than Wes Dooley's explanation, (which I have a hard time following). I'll try to post a complete answer by Sunday.

Chris Shaeffer :

I only have one mic that does figure 8 and I'm looking for some variety for M/S experimentation (my new toy). What kind of trouble am I setting up for with this idea? Has anyone tried it?

Using 2 cardioids set up as close as X/Y micing but facing away from each other to mimic figure 8 and using the combination as my side signal.

Something tells me it won't work well, but I don't know why it won't unless it has something to do with the omni-ness of cardioid mics with low freqs.

Harvey Gerst :

Breaking down the M/S technique in simple terms.

Okay, I'm gonna try and do this without vectors, math, "sum and difference" signal theory, and no pictures - except for the pictures we'll draw in our mind. Let's start by imagining a figure 8 ribbon mic and how it works, since the concept is really simple.

A figure 8 ribbon mic basically consists of a piece of thin metal ribbon, hanging in a large magnetic field. The ribbon is fasten at the top and bottom, and a small wire is connected to each end of the ribbon. When there is a sound, it causes the ribbon to move inside the magnetic field; a small voltage is generated, and that's the signal that comes down the cable.

The ribbon is free to move forward and backward, but not side to side. If a sound comes in from the front, it puts out a positive signal. If a sound comes in from the back, it puts out a negative signal. If a sound comes in from the side, it hits both the front and back of the ribbon equally, and no signal comes out.

Okay, now let's put this mic into our mixing board, set all the balance controls straight up, and listen to it.

If we put a singer in front of the mic and another singer in back of the mic, it'll pick up both singers pretty equally, because the front and back of the figure 8 ribbon mic are both open. A singer singing into the side of the mic won't be heard at all.

Okay, is everybody cool on all this stuff so far? Cuz here's where we get a little tricky.

Let's imagine 3 trumpet players spread out across a stage. The player on the left is A, in the center is B, and at the right is C.

At the same time, we're gonna turn our mic sideways, so that the front of the mic is facing the trumpet player (A) standing at the left side of the stage, the back of the mic is facing the trumpet player (C) standing at the right side of the stage, and the side of the mic is facing the trumpet player (B), standing in the middle of the stage. When we listen to the recording, we can hear A and C clearly, and we won't hear B hardly at all, as expected.

K, here's where the "sleight of hand" magic trick comes in.

Let's run the mic into a Y cable, except one of the ends of the Y cable is wired backwards. This means we'll have a normal positive signal going to one channel, AND a mirror image negative signal going to the other channel.

If you bring up the level of either channel by itself, you'll get a good sound. But if you leave one channel up and bring the other channel up slowly, the ENTIRE signal will eventually disappear when the levels are exactly equal. The plus and minus signals exactly cancel out. But here's where it gets wild.

Stick a cardioid mic on top of the figure 8 and point it towards the center of the stage, and bring up it's level on a third mixer channel, again with the pan control centered. We pretty much hear the middle trumpet now and we hear the two trumpets on both sides of the stage, but softer than the middle one. And the ribbon mic is still silent, so why even bother with it?

Here's the magic trick:

Move the pan control on the first ribbon mic's channel all the way to the left. Move the pan control on the backwards wired channel all the way to the right. Whoa, full, glorious stereo. Kill the middle mono mic and the entire signal disappears again. Bring back the mono center mic and we have full stereo again. What the hell is going on - what's happening?

Think of each trumpet as putting out a little puff of positive air. As the trumpet note from A hits the ribbon the mic faithfully records it to the track that corresponds to that ribbon mic's channel. At the same time it records a mirror image negative to the next ribbon track (since it's wired backwards). It also records a positive puff to the cardioid in the center).

When you add that center cardioid mic, it combines with the ribbon signal from the left, and moves the apparent sound all the way to the left. The trumpet on the right gets the reversed channel's attention and it combines with the center mic to produce a very right hand sounding signal. The center trumpet is only heard by the cardioid so it only comes bact thru the center.

By adjusting the level of the side channels relative to the center mic, you can get any stereo spread you want, from full stereo to perfect mono.

So can you use condenser mics that have a figure eight pattern, or three cardioids, each pointing in a different direction? Yes, as long as you can get the two side channels to null completely, or damn close to it. One mic or side has to face left and the other side/mic has to face right, and the two channels hafta cancel out almost completely. How you do that is up to you. The cardioid mic faces forward and completes the stereo information gathering.

That's how M/S works. I use/like ribbons for the side mic cuz they just have one moving part, no phantom power to worry about, and if the mixer channel has a polarity reverse switch, a simple Y cord will work. And the signal is automatically identical to start with.

Does this about cover it?

Chris Shaeffer :

OK, I may not have it exactly right, but it sure is fun! Here's what I did: GT 5sm in figure 8 into Art MP Channel 1 1/4" out to Track 1 in Cubase XLR out to Art MP Channel 2 with phase reversed 1/4" out to rack in Cubase C1 in the center into Track 3 in Cubase. Set levels on the MP so that they are identical (which was quite a game), set levels equally into the computer...test in mono- yup the sides pretty much disappear.

Play some Tom Petty-ish stuff from the center position. (Feel like an idiot...) Sounded like doo doo in the cans.

Back in Cubase, panned 1 & 2 hard left and right. Left 3 dead center. Left all levels where they were- roughly equal. Pressed play...

That's some WIDE stereo! And loud as heck. OK, hit the mono switch to see what it does....

Drops about 12dB. Is it supposed to do that?

Hmmm...reread Harvey's post several times.

I fiddled with the levels and found that if I keep the side channels lower (about -15dB) I can still get a nice full stereo image that isn't too affected when I sum to mono (and doesn't sound too much like my...ah..."studio" room). I'm not entirely sure what's going on, or even whether I have it right or wrong...

When I mute the center and have the sides panned out I get a really yucky, quiet, thin stereo. I was expecting it to cancel out alomst completely. If I then hit the mono switch it *almost* disappears- there is only a tiny bit left- proabably the inconsistencies in the two signal chains. The meters say it peaks at -22dB but I can hardly hear it.

When I bring the center back the stereo effect is much more noticeable than the tinny sides by themselves so I think its working the way it should, but I don't really have perfect mono compatibility because of my half-arsed signal chaining rig.

I wonder if:

- The figure 8 pattern on the GT isn't tight enough.

- My funky signal chain is causing this volume change in mono.

- It would work better recording more than one instrument.

- I somehow got it perfect and this is normal. (Not likely....)

Yehaw! Another fun evening courtesy of Harvey Gerst. Someday I might even understand some of what I'm doing.

Henrik :

A few questions, though:

What are the pros and cons of using the technique you describe above compared with using a matrix box or a preamp that can function as one (such as Joemeek VC7)? If you do the latter, can you control the amount of stereo spread after having recorded it, or are you stuck with the relative levels of the M and S mics?

If you connect a mic to a Y-cable, will it perform as well as it would with just one cable? Also, what are your opinions on the different ribbon mics mention? Any recommendations?

Harvey Gerst :

The matrix box means not having to make up special cables or worry about matching stuff. It just makes life a little easier.

Any good ribbon or figure 8 mic should do a pretty good job. The Coles 4038 is avery good choice and many people like the AKG C414 for M/S. The stereo spread is determined by the level of the M mic relative to the S signal.

Henrik :

Harvey, I'm still wondering - what are your reasons for preferring spaced omnis to M-S while recording drums?

Harvey Gerst :

Mainly because:

1. One of our drum rooms is too frigging small to really get a great natural drum sound in.

2. Most of the stuff I do is rock and metal, and spaced omnis or spaced cardioid miking gives me a few more options later during mixdown.

3. It makes the drummer feel more professional when I run 10 channels of drums for a 4 piece rock group. That leaves a channel for the vocal, a channel for bass, and 12 channels for the guitar, which works out about right these days.

Henrik :

A friend of mine just visited a studio where they were recording drums - using no less than 32 mics! Now what's THAT all about!? Well, maybe they were trying to make the drummer feel imoportant.

Anyway, I'm also really curious to hear what you think Chris Schaeffer did wrong when trying out the M-S technique (a few posts above).

Harvey Gerst :

Some engineers are very anal about miking drums. They mic each head (top AND bottom), each cymbal, and then throw some room mics in as well. Most engineers aren't usually that persnickity.

knownuttin :

What that's all about is massive phase cancellation. Unless it's just a placebo for the drummer and 20 or so aren't plugged into anything. Or maybe they like a bunch of different room mics!

Al Sim :

Thanks Harvey

This post is for the man, Harvey Gerst. I finally had a chance to use the Marshall MXL-603S and the Behringer ECM8000 that you reccommended on my friend's old J-45. The beast has been tamed. Many thanks and hats off to a true master.

wes480 :

Maybe this has been covered - but I havn't seen it. On the one hand, the 3:1 rule is reccomended... Each mic should be 3 times as far away from each other as the first one is to the source. So why does XY micing work? If the mics are so close. Seems like if say, one was 2 feet from the guitar...according to the 3:1 rule, the next mic would have to be like 6 feet away from that mic.

With XY...they are right next to each other - I don't get it

Harvey Gerst :

The 3:1 rule is really for when you're recording mono instruments, although it has some bearing in stereo (like drum overheads).

XY places the mic capsules so close together, that they essentially pick up exactly the same signal from an instrument directly in front of them, without phase problems. As the instruments spread out from the center, they are picked up by one mic more than the other. Because they are so close together, the phase problems are minimized.

You can also use two mics, if they are exactly equal distance from the source, with out phase problems. That's the basis behind ORTF, NOS, and several "near coincidence" stereo miking techniques.

wes480 :

a) So - why does XY miking sound better than just using 1 mic?

b) If they are basically pikcing up the same source....just so you can easily pan? is the only goal with that? makes sense i guess....

c) As far as 3:1 - "mono instruments"? Obviously that doesn't mean an instrument you are recording with just 1 mic....becuase then 3:1 wouldn't come into play, so..I am not sure what a "mono instrument" is.

d) In terms of equidistant from the source....is that like you said always have your overheads the exact same distant from the snare? On a drumset...

e) I havn't tried it yet..but it seems like that would really hurt the stereo image. You just need to bring one mic in closer than the other?

Harvey Gerst :

OK, I can see where my response might have been confusing so let's see if I can clear things up a bit

a) X/Y is a recording technique for getting a good stereo image of a wide sound source, using just two microphones.

b) Using multiple mics at different distances from a mono source is a technique for getting different tonal colors and interesting time delays that can add a distictive character to the sound that isn't possible with just one mic.

c) An electric guitar where the sound comes out of one speaker would be a good example of a "mono source". Any instrument that isn't going to have a stereo image in the final mix would be a mono source, even if you use several mics on the instrument. When you do use several mics on one instrument that will be blended together into one sound, you have to watch for phasing problems and that's where the 3:1 rule comes into play.

d) It's a good practice to follow, but sometimes you ignore the snare if you're trying to get more cymbal sounds and you use the overhead mics spread very wide. You have to listen for possible phasing problems, but if you mic the cymbals pretty close, it's not a big problem.

e) Since the snare is probably the main focus in a drum set, you always make sure that your multiple mic setup on drums isn't causing problems with the basic tone of the snare from mic phasing problems due to poor mic placement.

X/Y, ORTF, NOS, "near coincendence" miking are just techniques for getting a good stereo image of a wide source, using just two mics.

Whenever you go to more than just those two microphones (e.g., close miking a set of drums with 3 or more mics), you can create phasing problems caused by multiple mics picking up the same source from small differences in distance, which creates phasing problems.

wes480 :

a) So, in terms of acoustic instruments...the 3:1 rule isn't a big deal.

b) And, if you are recording an acoustic instrument with only 2 mics....then you really don't have to worry about phasing issues most of the time?

c) And XY miking is essentially just like using an AT "stereo mic" or something?

Harvey Gerst :

a) Yeah, it is a big deal if the mics aren't the same distance from the instrument; then the 3:1 rule comes into play.

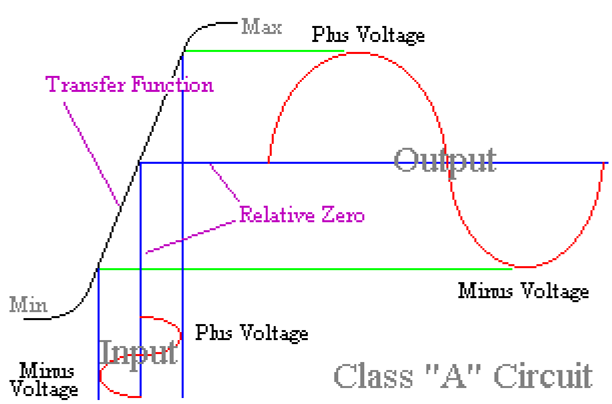
b) If the mics are in X/Y (or one of the standard stereo miking configurations), no, you don't hafta worry about phasing. If the mics are at different distances from the acoustic instrument, yes, you hafta worry about phasing.

c) Yeah, kinda like that.

Harvey Gerst :

Clearing up some misconceptions about tube mics and preamps.

OK, here we go about tubes. As far as most tube circuits in mics and preamps are concerned, they're usually a single tube operating as a "Class A" device. Here's a picture of how a Class A circuit works:



See the signal coming in, at the bottom of the drawing? It goes into the tube (which is represented by that "S" shaped curve), hits the transfer function and can come out amplified (as in the right side of the drawing), or simply come out equally, depending on the circuit designer's intent.

It's important to realize that the output of a condenser capsule is very low, while the capsule's impedance is VERY, VERY high. Very high impedance sources don't travel well over long distances, so it's important to convert that high impedance to a lower impedance as soon as possible.

You have two choices: a Field Effect Transistor (commonly called an FET), or a tube (some tubes love seeing very high impedance sources). If you use an FET, it won't do much in the way of giving you any more signal, so you'll need to add some more transistors to boost the signal a little bit. And most transistor circuits tend to distort very easily (and in a nasty way) if pushed too hard.

Tubes, on the other hand, tend to simply round off the signal if they approach the top and bottom of the "S" shaped "Transfer Curve", resulting in more musical distortion components, i.e., more 2nd and 4th harmonic overtones, which are musically correct, creating a "warmer, fatter tone. That's what makes tube distortion so desirable in guitar amplifiers.

Some tube preamp designs add more distortion by using a very small plate voltage to effectively shrink the length of that straight line part of the "Transfer Function" so that the tube saturates quicker and distorts faster. To me, it sounds a little fake and un-natural, but a lot of people seem to like it.

So the main advantages to using tubes in mics are: natural impedance convertor, which also works as a gain stage, limiter, and as an even order distortion generator, when pushed hard. One lessor known aspect of using a tube inside the body of a micrphone is that the heat from the tube helps drive out any moisture in the capsule when used in humid environments.

Since the tube must have heater and plate voltages supplied from an outboard power supply, it also makes sense to generate the 48 volt phantom power voltage from the power supply as well. This brings up another possibility when using dual membrame capsules for multiple polar patterns: continuously variable remote polar pattern selection from the power supply.

Remember earlier in this thread, we discussed how the different pressure gradient polar patterns are created by mixing the sound from two polar patterns; omni and figure 8? We can take that one step further since a dual membrane condenser mic is made into an omni by having both sapsules charged. Flip the polarity of the back capsule's signal and you have a figure 8.

As you continuously adjust the level and polarity of the back capsule, the mic will slowly change the polar patterns starting with omni, passing thru wide cardioid, sub cardioid, hyper-cardioid, and super-cardioid on it's way to figure 8.

If you are using one of these continuously variable polar pattern mics, it can be used to remotely change the tone and the amount of proximity effect of the mic as well. As you move from omni to figure 8, the proximity effect goes from almost none to maximum.

Many engineers will use the pattern selector switch as a tone control and ignore the different polar pattern choices for a particular singer, since the mic is used in a pretty absorbent situation; an iso booth, or a very dead room. for example.

Often, the decision to use a tube mic is mistakenly made to increase distortion, resulting in what some people describe as "tube warmth". In most modern mic designs, tubes are used for the performance reasons (listed above), not to add distortion, but to eliminate the often unpleasant distortions caused by poorly designed transistor mic circuits, which can often be described as harsh, edgy, brittle, etc.

One last point about LD mic design: a 1" wavelength corresponds to a frequency around 5 to 7 kHz. Ever wonder why all 1" capsules have a peak right around that frequency range? Now you know. Explaining what to do about it is a whole 'nother subject, which we'll get into at another time.

I hope some of this has proved helpful to at least somebody out there.

h kuhn :

Harvey, how would the drawing differ for a class A/B or class B circuit?

Harvey Gerst :

Class AB and B divide the signal up into two section: the positive half, and the negative half. A seperate tube drives each half of the signal and they're recombined in the output stage. Unless the cirduit is designed very carefully, right where each side shuts off (as it hits the zero point) can create a slight lag, causing what's called "crossover" or "notch" distortion. It's a very ugly sound.

Class AB trys to prevent this by having each side operate as Class A at very low levels (i.e., both sides passing the whole signal), gradually switching to class B as the signal gets louder and louder.

Class A is usually used in low level signal amplification (preamps, mics, etc.), whereas speaker amplifiers generally use Class AB, or other classes of amplification.

muzeman :

(Harvey) If you are using one of these continuously variable polar pattern mics, it can be used to remotely change the tone and the amount of proximity effect of the mic as well. As you move from omni to figure 8, the proximity effect goes from almost none to maximum

When you said change the tone and proximity effect, does this mean you can change the tonal character of the mic to match the singers voice as well as the room?

When looking for a tube mic,are there certian electronic components to look for,other than class A circuitry,and others to avoid?

Harvey Gerst :

Yes, by using the polar pattern selector to combine the two diaphragms in different ways, the frequency response will change quite a bit. If the room is not contributing to the sound (by using an iso booth, or close-in absorbtion baffles), you can use the pattern selector as a kind of weird tone control, since it alters several characteristics at the same time (like frequency response, angular response, and proximity effect). It gives you a lot more color variations to choose from.

muzeman :

Have you heard the T-3?

Harvey Gerst :

I sent all of Alan's mics to Stephen Paul for his comments, so I really didn't have a chance to put it up and really listen to it except briefly. What I heard I liked, but I really didn't have them here as long as I would have liked. If Stephen is finished with them, maybe he or Alan will send them back so I can do a more in-depth report on them.

muzeman :

Back to tubes,sorry to get off track.

a) Do you think there's a great advantage/disadvantage between class A verses A/B,B designs in tube mics and pres?

b) Unfourtanatly,most of us around here could never afford a class A tube mic or pre.

c) Would it be better to stick with a non tube mic or pre that's class A,than go for an AB/B tube design?

Harvey Gerst :

a) Pretty much most single tube devices that amplify have to be Class A, but there's good Class A design and bad Class A design. The other classes come into play when you need a lot of power and you want to split the work between two or more output devices, like a power amplifier.

b) Prices are dropping on these things, but again, there's good Class A design and bad Class A design.

c) There's no tube mic I know of that uses a Class A/B or B circuit. The best mics and pres will be the best sounding units that have well designed circuits and don't worry about the class - that's the designer's job - to figure out how to make the device sound good.

When people make these things, there are only two or three possible choices; Make them almost one at a time, using the best components possible, and sell them for enough money to make a decent profit. This is the way they build Manley, Brauner, Milinia Media, or when you send something to Stephen Paul for modifications.

Or you can setup a factory, hire knowledgeable people, use good components, build them very well, and sell a lot of them at a lower price.

The last way to lower your cost and lower the price is to have somebody else build something in a place that has very cheap labor and is set up to crank out a ton of products. In order to do that, you need to install a large amount of QC to insure the product does what it's supposed to and that the outside supplier hasn't screwed up somewhere.

Grotius :

Wow, I've just spent three hours reading this thread; Harvey, this is a single best thing I've read on microphones anywhere. Thank you so much for educating me. I'm sort of nervous posting in this famous thread because I'm a relative newbie, but what the heck; I have three questions:

a) I noticed that Harvey recommended a (matched pair?) of small diaphragm omni or cardioid condensers for recording classical grand piano, which is my main interest. Noise and dynamic range are obviously important to me. My main question is about the relative importance of various specifications, especially self-noise and SPL. Is low self-noise more important than high maximum SPL? (FYI, I have a new 7-foot Bluthner in my living room, and my tastes lean toward "quieter" classical music like Chopin and Debussy, though I do enjoy playing the occasional Beethoven and Rachmaninoff.)

b) I'm also curious whether Harvey (or anyone else) has a preference among specific microphones for this application. I've heard various recommendations, including the KM184, KM183, TLM103, Rode NT2, AT 4033, AT822, ATM25, ATM87R, Oktava MC012s, and MXL 2003. Or, er, the DPA 4011, if one can afford it (which I can't). Also, in the low-self-noise department, does the TLM103 take the crown?

c) Finally, I gather that a single stereo pair is enough? Too many phase problems if one tries 4 mikes, e.g. to close-mike and distance-mike at the same time?

Harvey Gerst :

a) Dynamic range isn's as much of a problem as self noise when it comes to recording classical piano.

b) The ATM25 should be removed from your list. The Neumanns are pretty standard for miking piano, but they have a tendency to be a little bright. A good pair of Soundroom MC012s or even the Marshall MXL 603S might work for you. All of the mics you list are pretty good for piano, but finding the best matchup for your piano and playing style will take some work.

c) A stereo pair will usually do the job, although adding an extra mic for the room is often a nice way to get real ambience.

Chris F :

The best sound I've ever gotten on my grand (1937 Baldwin 6'3") came from an Earthworks omni, but that was borrowed and I can't afford the price tag. I get good results with a pair of MXL 603's in the x/y configuration, although you'll have to do a lot of testing to figure out what kind of sound you're after. You'll get a very powerful sound if you go close to the strings, but you'll also hear the mechanism of the action and the "whoosh" of the dampers every time you pedal when you do this. Each placement scheme gives a slightly different color, and only your ears can tell you what kind of sound suits your taste.

Two things to be careful of: a bright piano (or an old one with old hammers) brings all of the imperfections of the instrument to the forefront of the recording, and you can hear some wolf-tones (overtones, etc.) on the playback that you never even knew existed. Also, if your action hasn't been regulated and your strings leveled, you might be in for some strange surprises as the mics are able to pick up a level of detail that your ears don't usually catch from further away.